



Synway SBO Series Gateway

SBO500 Gateway

User Manual

Version 1.6.2

Synway Information Engineering Co., Ltd

www.synway.net

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Revision History

Version	Date	Comments
Version 1.0.0	2015-05	Initial publication.
Version 1.6.2	2015-09	New revision

Note: Please visit our website <http://www.synway.net> to obtain the latest version of this document.

Chapter 1 Product Introduction

Thank you for choosing Synway SBO Series Gateway!

The Synway SBO series gateway products (hereinafter referred to as 'SBO gateway') are mainly used for connecting IP or enterprise PBX with the IP telephony network or IP PBX. It provides such functions as transcoding, routing, number filtration, number manipulation and so on. Currently, only SBO500 is available for you.

1.1 Typical Application

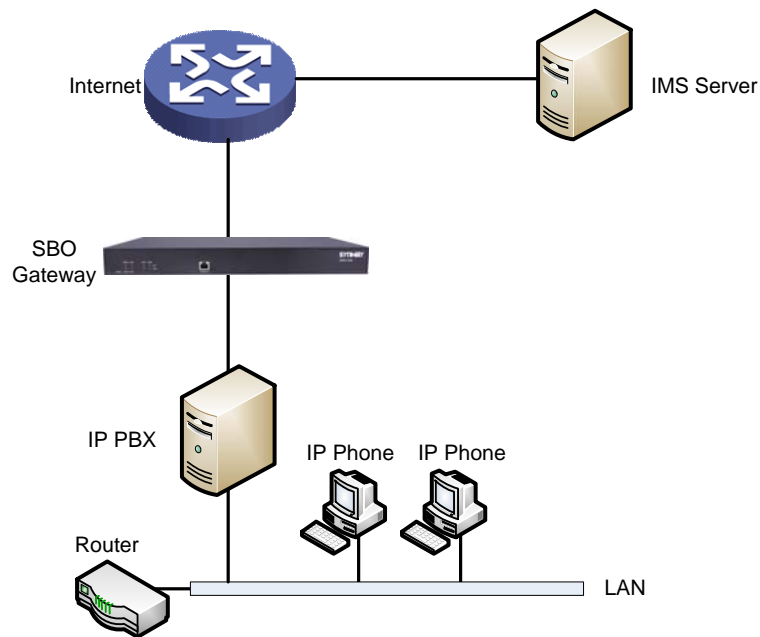


Figure 1-1 Typical Application

1.2 Feature List

Basic Features	Description
IP Call	Call initiated from IP to a designated SIP trunk, via routing and number manipulation.
Number Manipulation	Peels off some digits of a phone number from left/right, or adds a prefix/suffix to a phone number.
VoIP Routing	Routing path: from IP to IP.
Signaling & Protocol	Description
SIP Signaling	Supported protocol: SIP V1.0/2.0, RFC3261
Voice	CODEC G.711A, G.711U, G.729, G723, G722, AMR, iLBC DTMF Mode RFC2833, SIP INFO, INBAND
Network	Description
Network Protocol	Supported protocol: TCP/UDP, HTTP, ARP/RARP, DNS, NTP, TFTP, TELNET, STUN
Static IP	IP address modification support
DNS	Domain Name Service support
Security	Description
Admin Authentication	Support admin authentication to guarantee the resource and data security
Maintain & Upgrade	Description
WEB Configuration	Support of configurations through the WEB user interface
Language	Chinese, English
Software Upgrade	Support of user interface, gateway service, kernel and firmware upgrades based on WEB
Tracking Test	Support of Ping and Tracert tests based on WEB
SysLog Type	Three options available: ERROR, WARNING, INFO

1.3 Hardware Description

The SBO gateway features 1U rackmount design and integrates embedded LINUX system within the POWERPC+DSP hardware architecture. It has 2 Kilomega-Ethernet ports (NET1 and NET2) on the chassis. See the figures below for SBO500 appearance:



Figure 1-2 Front View



Figure 1-3 Rear View



Figure 1-4 Left View

The table below gives a detailed introduction to the interfaces, buttons and LEDs illustrated above:

Interface	Description
NET	Amount: 2
	Type: RJ-45
	Bandwidth: 10/100/1000Mbps
	Self-Adaptive Bandwidth Supported
	Auto MDI/MDIX Supported
Console Port	Amount: 1
	Type: RS-232
	Baud Rate: 115200 bps
	Connector: RJ45 (See Figure 1-5 for signal definition)
	Data Bits: 8 bits
	Stop Bit: 1 bit
	Parity Unsupported
	Flow Control Unsupported
Button	Description

Power Key	Power on/off the SBO gateway. You can turn on the two power keys at the same time to have the power supply working in the hot-backup mode.
Reset Button	Restore the gateway to factory settings.
LED	Description
Power Indicator	Indicates the power state. It lights up when the gateway starts up with the power cord well connected.
Run Indicator	Indicates the running status. For more details, refer to 1.4 Alarm Info .
Alarm Indicator	Alarms the device malfunction. For more details, refer to 1.4 Alarm Info .
Link Indicator	The green LED on the left of NET, indicating the network connection status.
ACT Indicator	The orange LED on the right of NET, whose flashing tells data are being transmitted.

Note: The console port is used for debugging. While connection, the transmitting and receiving lines of the gateway and the remote device should be cross-linked. That is, connect the transmitting line of the gateway to the receiving line of the remote device, and vice versa. The figure below illustrates the signal definition of the console port on the gateway.

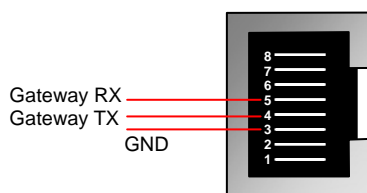


Figure 1-5 Console Port Signal Definition

For other hardware parameters, refer to [Appendix A Technical Specifications](#).

1.4 Alarm Info

The SBO gateway is equipped with two indicators denoting the system's running status: Run Indicator (green) and Alarm Indicator (red). The table below explains the states and meanings of the two indicators.

LED	State	Description
Run Indicator	Go out	System is not yet started.
	Light up	System is starting.
	Flash	Device is running normally.
Alarm Indicator	Go out	Device is working normally.
	Light up	Upon startup: Device is running normally. In runtime: Device goes abnormal.
	Flash	System is abnormal.

Note:

- The startup process consists of two stages: System Booting and Gateway Service Startup. The system booting costs about 1 minute and once it succeeds, both the run indicator and the alarm indicator light up. Then after the gateway service is successfully started and the device begins to work normally, the run indicator flashes and the alarm indicator goes out.
- During runtime, if the alarm indicator lights up or flashes, it indicates that the device goes abnormal. If you cannot figure out and solve the problem by yourself, please contact our technicians for help. Go to [Appendix C Technical/sales Support](#) to find the contact way.

Chapter 2 Quick Guide

This chapter is intended to help you grasp the basic operations of the SBO gateway in the shortest time.

Step 1: Confirm that your packing box contains all the following things.

- SBO Series Gateway *1
- Angle Bracket *2, Rubber Foot Pad *4, Screw for Angle Bracket *8
- 220V Power Cord *2
- Warranty Card *1
- Installation Manual *1

Step 2: Properly fix the SBO gateway.

If you do not need to place the gateway on the rack, simply fix the 4 rubber foot pads. Otherwise, you should first fix the 2 angle brackets onto the chassis and then place the chassis on the rack.

Step 3: Connect the power cord.

Make sure the device is well grounded before you connect the power cord. Check if the power socket has the ground wire. If it doesn't, use the grounding stud on the rear panel of the device (See Figure 1-3) for earthing.

Note: Each SBO gateway has two power interfaces to meet the requirement for power supply hot backup. As long as you properly connect and turn on these two power keys, either power supply can guarantee the normal operation of the gateway even if the other fails.

Step 4: Connect the network cable.

Step 5: Log in the gateway.

Enter the original IP address (NET 1: 192.168.1.101 or NET 2: 192.168.0.101) of the SBO gateway in the browser to go to the WEB interface. The original username and password of the gateway are both 'admin'. For detailed instructions about login, refer to [3.1 System Login](#). We suggest you change the initial username and password via 'System Tools → Change Password' on the WEB interface as soon as possible after your first login. For detailed instructions about changing the password, refer to [3.7.14 Change Password](#). After changing the password, you are required to log in again.

Step 6: Modify IP address of the gateway.

You can modify the IP address of the gateway via 'System Tools → Network' on the WEB interface to put it within your company's NET. Refer to [3.7.1 Network](#) for detailed instructions about IP modification. After changing the IP address, you shall log in the gateway again using your new IP address.

Step 7: Check the channel status.

You can check the status of the channels via 'Operation Info → IP Status'. Refer to [3.2.2 IP Status](#) for detailed introductions.

Step 8: Set routing rules for calls.

Note: For your easy understanding and manipulation, all examples given in this step do not involve registration.

Step 1: Configure the IP address of the remote SIP terminal which can establish conversations with the gateway so that the calls from other terminals will be ignored. Refer to 'SIP Settings → [SIP Trunk](#)' for detailed instructions. Click **Add New** to add a new SIP trunk,

fill in 'Remote IP' and 'Remote Port' with the IP address and port of the remote SIP terminal which will initiate calls to the gateway. You may use the default values for the other configuration items.

Example: Provided the incoming IP address of the SIP trunk is 192.168.0.111 and the port is 5060. Add **SIP Trunk 0**; set **Remote IP** to **192.168.0.111** and **Remote Port** to **5060**. The outgoing IP address of the SIP trunk is 192.168.0.222 and the port is 5060. Add **SIP Trunk 1**; set **Remote IP** to **192.168.0.222** and **Remote Port** to **5060**.

Step 2: Add the IP address of the SIP trunks configured in Step 1 into the corresponding SIP trunk group. Refer to SIP Settings → [SIP Trunk Group](#) for detailed instructions. Click **Add New** to add the SIP trunk group. Select the SIP trunk configured in Step 1 as 'SIP Trunks'. You may use the default values for the other configuration items.

Example: Add **SIP Trunk Group 0**. Check the checkbox before **0** for **SIP Trunks** and keep the default values for the other configuration items. Add **SIP Trunk Group 1**. Check the checkbox before **1** for **SIP Trunks** and keep the default values for the other configuration items.

Step 3: Add routing rules. Refer to 'Route Settings → [IP→IP](#)' for detailed instructions. Select the SIP trunk group 0 set in Step 2 as 'Call Initiator' and the SIP trunk group 1 set in Step 2 as 'Call Destination'. You may use the default values for the other configuration items.

Example: Select **SIP Trunk Group[0]** as **Call Initiator** and **SIP Trunk Group[1]** as **Call Destination**. Keep the default values for the other configuration items.

Step 4: Initiate a call from the SIP trunk 0 configured in Step 1 to the IP address and port of the SBO gateway. Thus you can establish a call conversation via SIP trunk[1] with the IP terminal. (Note: The format used for calling an IP address via SIP trunk is as follows: username@IP address.)

Example: Provided the IP address of the SBO gateway is 192.168.0.101 and the port is 5060. Provided 123 is a number which conforms to the number receiving rule of the remote device. Initiate a call from SIP trunk 0 to the IP address 192.168.0.101 (in the format: 123@192.168.0.101) and you can establish a call conversation via SIP trunk[1] to the number 123.

Special Instructions:

- The chassis of the SBO gateway must be grounded for safety reasons, according to standard industry requirements. A simple way is earthing with the third pin on the plug or the grounding studs on the machine. No or improper grounding may cause instability in operation as well as decrease in lightning resistance.
- As the device will gradually heat up while being used, please maintain good ventilation to prevent sudden failure, ensuring that the ventilation holes (see Figure 1-4) are never jammed.
- During runtime, if the alarm indicator lights up or flashes, it indicates that the device goes abnormal. If you cannot figure out and solve the problem by yourself, please contact our technicians for help. Otherwise it may lead to a drop in performance or unexpected errors.

Chapter 3 WEB Configuration

3.1 System Login

Type the IP address into the browser and enter the login interface. See Figure 3-1.

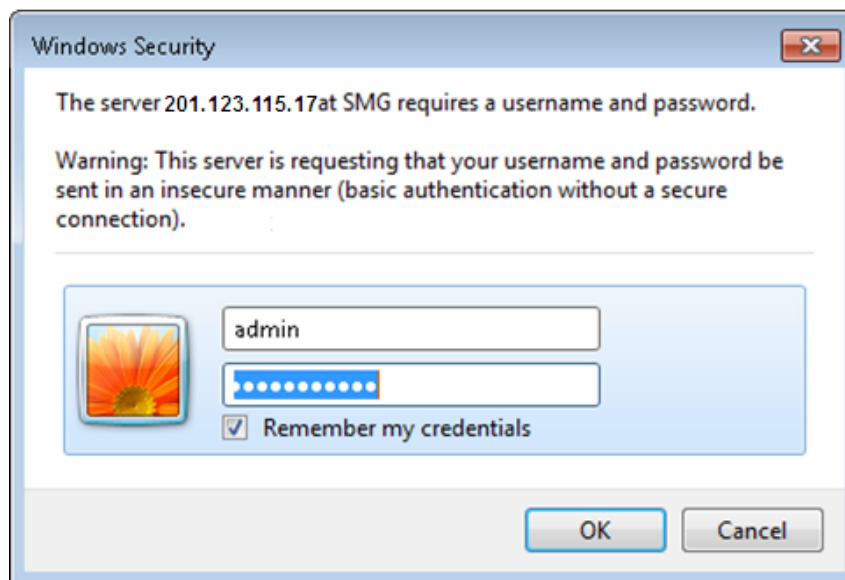


Figure 3-1 Login Interface

The gateway only serves one user, whose original username and password are both 'admin'. You can change the username and the password via 'System Tools → Change Password' on the WEB interface. For detailed instructions, refer to [3.7.14 Change Password](#).

After login, you can see the main interface as below.



Figure 3-2 Main Interface

3.2 Operation Info

Operation Info includes three parts: **System Info**, **IP Status** and **Call Count**, showing the current running status of the gateway. See Figure 3-3.

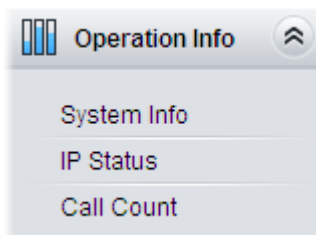


Figure 3-3 Operation Info

3.2.1 System Info

System Info			
LAN 1			
MAC Address	80:7B:85:10:04:92		
IP Address	201.123.112.17	255.255.255.0	201.123.112.254
DNS Server	0.0.0.0		
Receive Packets	All:176962	Error:0	Drop:0
Transmit Packets	All:695	Error:0	Drop:0
Current Speed	Receive:1.3 KB/s	Transmit:638 B/s	
Work Mode	100Mb/s Full Duplex		
LAN 2			
MAC Address	80:7B:85:10:04:93		
IP Address	192.168.0.101	255.255.255.0	192.168.0.254
DNS Server	0.0.0.0		
Receive Packets	All:0	Error:0	Drop:0
Transmit Packets	All:0	Error:0	Drop:0
Current Speed	Receive:0 B/s	Transmit:0 B/s	
Work Mode	Unknown! Unknown		
Runtime	3h 40m 37s		
Current Version			
Serial Number	000001861		
WEB	1.6.2_2015082016		
Gateway	1.6.2_2015082016		
Uboot	2.0.9_201505		
Kernel	#351 SMP Wed Aug 5 10:15:28 CST 2015		
Firmware	16		
Refresh			




































































Figure 3-4 System Info Interface








See Figure 3-4 for the system info interface. You can click **Refresh** to obtain the latest system information. The table below explains the items shown in Figure 3-4.

Item	Description
------	-------------

MAC Address	MAC address of NET 1 or NET 2.
IP Address	The three parameters from left to right are IP address, subnet mask and default gateway of NET 1 or NET 2.
DNS Server	DNS server address of NET 1 or NET 2.
Receive Packets, Transmit Packets	The amount of receive/transmit packets after the gateway's startup, including three categories: All, Error and Drop.
Current Speed	The current speed of data receiving and transmitting.
Work Mode	The work mode of the network, including five options: 10 Mbps Half Duplex, 10 Mbps Full Duplex, 100 Mbps Half Duplex, 100 Mbps Full Duplex and 1000 Mbps Full Duplex.
Runtime	Time of the gateway keeping running normally after startup. This parameter updates every 2s.
Serial Number	Unique serial number of a SBO gateway.
WEB	Current version of the WEB interface.
Gateway	Current version of the gateway service.
Uboot	Current version of Uboot.
Kernel	Current version of the system kernel on the gateway.
Firmware	Current version of the firmware on the gateway.

3.2.2 IP Status

Status	Idle	Ringing	Wait Answer	Dialing	Talking	Pending	Wait Message																											
Icon																																		
IP Status																																		
Channel No.	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31		
Channel Group1																																		
Channel Group2					<div>Talking</div> <div>Direction:IP Call In</div> <div>Caller:333</div> <div>Called:114</div>																													
Channel Group3																																		
Channel Group4																																		
Channel Group5																																		
Channel Group6																																		
Channel Group7																																		
Channel Group8																																		
Channel Group9																																		
Channel Group10																																		
Channel Group11																																		
Channel Group12																																		
Channel Group13																																		

	<i>Idle</i>		The channel is available.
	<i>Wait Answer</i>		The channel receives the ringback tone and is waiting for the called party to pick up the phone.
	<i>Ringing</i>		The channel is in the ringing state.
	<i>Talking</i>		The channel is in a conversation.
	<i>Pending</i>		The channel is in the pending state
	<i>Dialing</i>		The channel is dialing.
	<i>Wait Message</i>		The channel is waiting for the message from remote PBX.

Note: The gateway provides the fuzzy search feature on this interface. After you click any characters on Figure 3-5 and press the 'F' button, the search box will emerge on the right top of this page. Then you can input the key characters and the gateway will locate the channel on which there is an ongoing call that conforms to the fuzzy search condition.

Take an example: As shown in Figure 3-6, after we input the character 114 to the search box, and click the **Search** button, the gateway does a fuzzy search and locates that the ongoing call whose CalledID contains the character 114 occurs on Channel 4 of Channel Group 1.

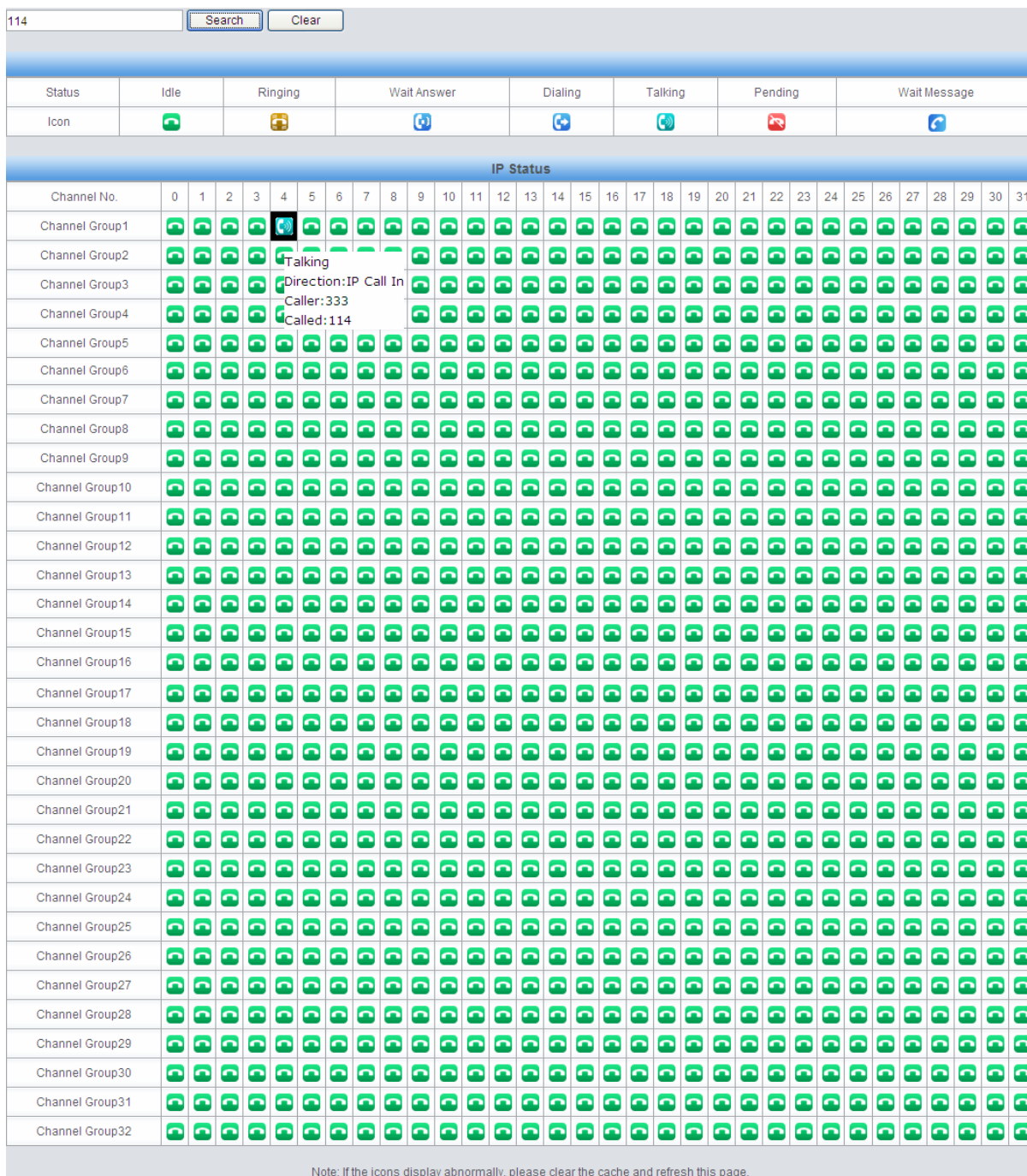


Figure 3-6 Search Calls

3.2.3 Call Count



Figure 3-7 Call Count Interface

See Figure 3-7 for the call count Interface. The above list shows the detailed information about all the calls counted from the startup of the gateway service to the latest open or refresh of this

interface. You can click **Refresh** to obtain the latest call count information, click **Download** to download the call count logs. The table below explains the items shown in Figure 3-7.

Item	Description
Release Cause	Reason to release the call.
Normal call clearing	Total number of the calls which are normally cleared.
Cancelled by calling party	Total number of the calls which are cancelled by the calling party.
User busy	Total number of the calls which fail as the called party has been occupied and replies a busy message.
No answer from user	Total number of the calls which fail as the called party does not pick up the call in a long time or the calling party hangs up the call before the called party picks it up.
Routing failed	Total number of the calls which fail because no routing rules are matched.
Resource unavailable	Total number of the calls which fail because no voice channel is available.
Call failed	Total number of the calls which fail as the called party number does not conform to the number-receiving rule or for relative reasons.
Others	Total number of the calls which fail because of other reasons.
Number	Total number of the calls on each state.
Percentage	The percentage of the calls with a release cause to total calls.

3.3 SIP Settings

SIP Settings includes six parts: **SIP**, **SIP Trunk**, **SIP Register**, **SIP Account**, **SIP Trunk Group** and **Media**. See Figure 3-8. **SIP** is used to configure the general SIP parameters; **SIP Trunk** is used to set the basic and register information of the SIP trunk; **SIP Register** is used for the registration of SIP; **SIP Account** is used for registering SIP accounts to the SIP server; **SIP Trunk Group** is to manage SIP trunks by group; and **Media** is to set the RTP port and the payload type.

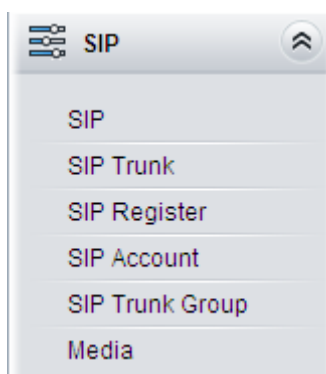


Figure 3-8 SIP Settings

3.3.1 SIP Settings

SIP Settings	
SIP Address of WAN	LAN 1: 201.123.112.17
Auto Change Default Gateway	<input type="checkbox"/> Enable
SIP Signaling Port	5060
Obtain CallerID from	Username of From Field
Obtain CalleeID from	Request Field
NAT Traversal	<input checked="" type="checkbox"/> Enable
Traversal Type	Port Mapping
LAN1 Mapping Address	
LAN2 Mapping Address	
SIP Transport Protocol	UDP
SIP Encryption	<input checked="" type="checkbox"/> Enable
Encryption Criterion	VOS1.1
Key	
RTP Encryption	<input type="checkbox"/> Enable
RTP Self-adaption	<input type="checkbox"/> Enable
UDP Header Checksum	<input checked="" type="checkbox"/> Enable
Rport	<input type="checkbox"/> Enable
DSCP	<input checked="" type="checkbox"/> Enable
Voice Media	46
Signal Control	26
Calls from SIP Trunk Address only	<input type="checkbox"/> Enable
Switch Signal Port if SIP Registration Failed	<input type="checkbox"/> enable
Working Period	<input type="checkbox"/> 24 Hours
Period(hh:mm:ss)	08:00:00 - 11:30:00
Maximum Wait Answer Time (s)	60
Maximum Wait RTP Time (s)	0

Figure 3-9 SIP Settings Interface

See Figure 3-9 for the SIP settings interface where you can configure the general SIP parameters. After configuration, click **Save** to save your settings into the gateway or click **Reset** to restore the configurations. If a dialog box pops up after you save your settings asking you to restart the service, do it immediately to apply the changes. Refer to [3.7.16 Restart](#) for detailed instructions. The table below explains the items shown in Figure 3-9.

Item	Description
SIP Address of WAN	IP address of SIP signaling for WAN, using NET 1 by default.
Auto Change Default Gateway	The SIP address of WAN will automatically shift to another LAN if the default one is unavailable. By default, the feature is disabled.
SIP Signaling Port	Monitoring port of SIP signaling. Range of value: 5001~65535, with the default value of 5060.
Obtain CallerID from	There are two optional ways to obtain the calling party number: from <i>Username of "From" Field</i> or from <i>Displayname of "From" Field</i> . The default value is from <i>Username of "From" Field</i> .
Obtain CalleeID from	There are two optional ways to obtain the called party number: from <i>"To" Field</i> or from <i>"Request" Field</i> . The default value is from <i>"Request" Field</i> .
NAT Traversal, Traversal Type	Sets whether to enable the NAT traversal. By default this feature is disabled. There is only one traversal type: <i>Port Mapping</i> .
LAN1 Mapping Address, LAN2 Mapping Address	The mapping addresses of LAN1 and LAN2 in case the NAT traversal is enabled. If the port mapping is selected as the traversal type, you are required to set the mapping address and port on the router and fill in the corresponding information here as well.
SIP Transport Protocol	There are two modes <i>UDP</i> and <i>TCP</i> available for running the SIP protocol. The default value is <i>UDP</i> .
SIP Encryption	Once this feature is enabled, you can encrypt the SIP signal following selecting an encryption criterion and setting a key. By default it is <i>disabled</i> .
Encryption Criterion	The criterion used to encrypt the SIP signal. At present only VOS1.1 is supported.
Key	The key to encrypt the SIP signal.
RTP Encryption	Once this feature is enabled, you can encrypt the RTP package. By default it is <i>disabled</i> .
RTP Self-adaption	When this feature is enabled, the RTP reception address or port carried by the signaling message from the remote end, if not consistent with the actual state, will be updated to the actual RTP reception address or port. By default, this feature is <i>disabled</i> .
UDP Header Checksum	When this feature is enabled, the gateway will automatically calculate the check sum of the UDP header during RTP transmission.
Rport	When this feature is enabled, a corresponding Rport field will be added to the Via message of SIP. By default, it is <i>disabled</i> .
DSCP	Sets whether to enable the DSCP differentiated services code point. By default, it is <i>disabled</i> .
Voice Media	Sets the priority of the voice media for DSCP. The voice media with a bigger value has a higher priority. The value range is 0~63, with the default value of 46.

Signal Control	Sets the priority of the signal control for DSCP. The signal control with a bigger value has a higher priority. The value range is 0~63, with the default value of 26.
Calls from SIP Trunk Address only	Once this feature is enabled, the gateway will only accept the calls from the IP addresses set in SIP Settings → SIP Trunk. By default, it is <i>disabled</i> .
Switch Signal Port if SIP Registration Failed	If the SIP registration fails, the SIP signaling port N will switch to N+1 for a new registration. It will continue until the registration succeeds.
Working Period, Period	The work period for the gateway. You can specify a certain period for the gateway to make calls. By default, the gateway is allowed to make calls any time in the day (24 Hours).
Maximum Wait Answer Time	Sets the maximum time for the SIP channel to wait for the answer from the called party of the outgoing call it initiates. If the call is not answered within the specified time period, it will be canceled by the channel automatically. The default value is 60, calculated by s.
Maximum Wait RTP Time	Sets the maximum time for the SIP channel to wait for the RTP packet. If no RTP packet is received within the specified time period, the channel will enter the pending state automatically and release the call. The default value is 0, calculated by s.

3.3.2 SIP Trunk

By default, there is no available SIP trunk information. Click **Add New** to add a new SIP trunk. See Figure 3-10 for the SIP trunk adding interface.

SIP Trunk

Index:

0

Remote Address:

Remote Port:

5060

WAN Option:

NET 1

Transport Protocol:

UDP

Outgoing Voice Resource:

512

Incoming Voice Resource:

512

Working Period:

☒ 24 Hours

Check	Codec	Check	Codec
<input checked="" type="checkbox"/> 1	G711A	<input checked="" type="checkbox"/> 5	iLBC
<input checked="" type="checkbox"/> 2	G711U	<input checked="" type="checkbox"/> 6	G723
<input checked="" type="checkbox"/> 3	G729	<input checked="" type="checkbox"/> 7	AMR
<input checked="" type="checkbox"/> 4	G722		

Save

Close

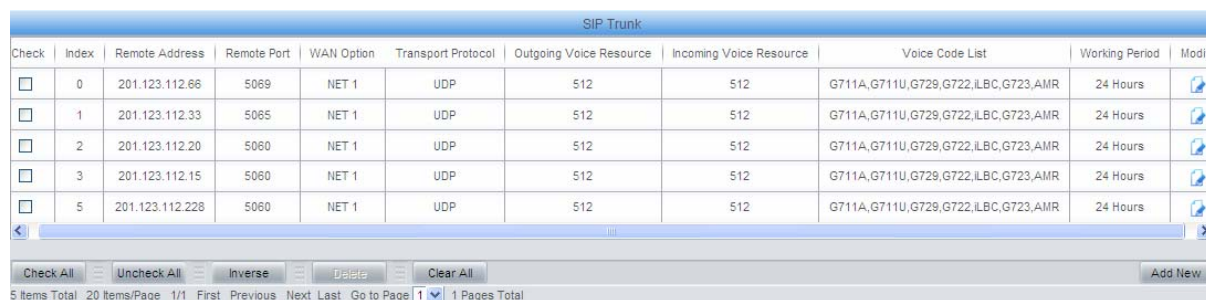
Figure 3-10 Add New SIP Trunk






The table below explains the items shown in Figure 3-10.

Item	Description
Index	The unique index of each SIP trunk.
Remote Address	Address of the SIP trunk, i.e. the IP address or domain name of the remote SIP terminal which will establish call conversation with the gateway.
Remote Port	Port of the SIP trunk.
Wan Option	Select the network port which used for WAN. The default setting is NET 1.
Transport Protocol	SIP transport protocol, providing two modes <i>UDP</i> and <i>TCP</i> . The default value is <i>UDP</i> .
Max. Voice Channels	Maximum number of voice channels allocated by the SIP trunk to the gateway.
Outgoing Voice Resource	Maximum number of voice channels for the outgoing calls allocated by the SIP trunk to the gateway.

Incoming Voice Resource	Maximum number of voice channels for the incoming calls allocated by the SIP trunk to the gateway.						
Working Period, Period	The work period for the gateway, You can specify a certain period for the gateway to make calls. By default, the gateway is allowed to make calls any time in the day (24 Hours).						
CODEC Priority	<p>Supported CODECs and their corresponding priorities for the SIP trunk to establish a call conversation. The table below explains the sub-items:</p> <table border="1"> <thead> <tr> <th>Sub-item</th><th>Description</th></tr> </thead> <tbody> <tr> <td>Priority</td><td>Priority for choosing the CODEC in an SIP conversation. The smaller the value is, the higher the priority will be.</td></tr> <tr> <td>CODEC</td><td>Seven optional CODECs are supported: G711A, G711U, G729, G722, G723, AMR and iLBC.</td></tr> </tbody> </table> <p>See 3.3.6 Media Settings for the detailed parameters for each CODEC.</p> <p>The default CODEC for the SIP trunk is the same as that set in 3.3.6 Media Settings.</p>	Sub-item	Description	Priority	Priority for choosing the CODEC in an SIP conversation. The smaller the value is, the higher the priority will be.	CODEC	Seven optional CODECs are supported: G711A, G711U, G729, G722, G723, AMR and iLBC.
Sub-item	Description						
Priority	Priority for choosing the CODEC in an SIP conversation. The smaller the value is, the higher the priority will be.						
CODEC	Seven optional CODECs are supported: G711A, G711U, G729, G722, G723, AMR and iLBC.						

After configuration, click **Save** to save the settings into the gateway or click **Close** to cancel the settings. See Figure 3-11 for the SIP Trunk Settings Interface



Check	Index	Remote Address	Remote Port	WAN Option	Transport Protocol	Outgoing Voice Resource	Incoming Voice Resource	Voice Code List	Working Period	Modify
<input type="checkbox"/>	0	201.123.112.66	5069	NET 1	UDP	512	512	G711A,G711U,G729,G722,iLBC,G723,AMR	24 Hours	
<input type="checkbox"/>	1	201.123.112.33	5065	NET 1	UDP	512	512	G711A,G711U,G729,G722,iLBC,G723,AMR	24 Hours	
<input type="checkbox"/>	2	201.123.112.20	5060	NET 1	UDP	512	512	G711A,G711U,G729,G722,iLBC,G723,AMR	24 Hours	
<input type="checkbox"/>	3	201.123.112.15	5060	NET 1	UDP	512	512	G711A,G711U,G729,G722,iLBC,G723,AMR	24 Hours	
<input type="checkbox"/>	5	201.123.112.228	5060	NET 1	UDP	512	512	G711A,G711U,G729,G722,iLBC,G723,AMR	24 Hours	

Check All Uncheck All Inverse Delete Clear All Add New

5 Items Total 20 Items/Page 1/1 First Previous Next Last Go to Page 1 1 Pages Total

Figure 3-11 SIP Trunk Settings Interface

Click **Modify** in Figure 3-11 to modify a SIP trunk. See Figure 3-12 for the SIP trunk modification interface. The configuration items on this interface are the same as those on the **Add New SIP Trunk** interface.

Check	Codec	Check	Codec
<input checked="" type="checkbox"/> 1	G711A	<input checked="" type="checkbox"/> 5	iLBC
<input checked="" type="checkbox"/> 2	G711U	<input checked="" type="checkbox"/> 6	G723
<input checked="" type="checkbox"/> 3	G729	<input checked="" type="checkbox"/> 7	AMR
<input checked="" type="checkbox"/> 4	G722		

Figure 3-12 Modify SIP Trunk

To delete a SIP trunk, check the checkbox before the corresponding index in Figure 3-11 and click the **Delete** button. **Check All** means to select all available items on the current page; **Uncheck All** means to cancel all selections on the current page; **Inverse** means to uncheck the selected items and check the unselected. To clear all SIP trunks at a time, click the **Clear All** button in Figure 3-11.

3.3.3 SIP Register

By default, there is no SIP register available on the gateway. Click **Add New** to add them manually. See Figure 3-13.

Figure 3-13 Add SIP Register Interface

The table below explains the items shown in the above figure.

Item	Description
Index	The unique index of each SIP register.
SIP Trunk No.	The number of the SIP trunk which registers to the SIP server.
Username	When the gateway initiates a call to SIP, this item corresponds to the username of SIP; when the gateway initiates a call to IP, this item corresponds to the displayed CallerID.
Password	Registration password of the gateway. To register the gateway to the SIP server, both configuration items Username and Password should be filled in.
Register Address	Address of the SIP server to which the SIP trunk is registered.
Register Port	The signaling port of the SIP trunk.
Domain Name	Domain name of the gateway used for SIP registry.

Register Expires	Validity period of the SIP registry. Once the registry is overdue, the gateway should be registered again. Range of value: 10~3600, calculated by s, with the default value of 3600.
IMS Network	Once this feature is enabled, the gateway will send signaling messages to the corresponding externally bound address and port when it registers to the server. Only when this feature is <i>enabled</i> will these items Externally Bound Address , Externally Bound Port and Authentication Username be shown.
Externally Bound Address	Externally bound IP address for registration.
Externally Bound Port	Externally bound port for registration.
Authentication Username	Authentication username for registration.

After configuration, click **Save** to save the settings into the gateway or click **Close** to cancel the settings. See Figure 3-14 for the SIP Register Information List.



Check	Index	SIP Trunk No.	Username	Register Address	Register Port	Domain Name	Register Expires (s)	Register Status	IMS Network	Externally Bound Address
<input type="checkbox"/>	0	0	100	201.123.115.26	5060	--	3600	Failed	No	--

Check All Uncheck All Inverse Delete Clear All Add New

1 Items Total 20 Items/Page 1/1 First Previous Next Last Go to Page 1 1 Pages Total

Figure 3-14 SIP Register Information List

Click **Modify** in Figure 3-14 to modify a SIP register. The configuration items on the SIP Register Modification Interface are the same as those on the **Add New SIP Register** interface.

SIP Register

Index:

SIP Trunk No.: ▼

Username:

Password:

Register Address:

Register Port:

Domain Name:

Register Expires (s):

IMS Network: ▼

Externally Bound Address:

Externally Bound Port:

Authentication Username:

Save

Close

Figure 3-15 SIP Register Modification Interface

To delete a SIP register, check the checkbox before the corresponding index in Figure 3-14 and click the **Delete** button. **Check All** means to select all available items on the current page; **Uncheck All** means to cancel all selections on the current page; **Inverse** means to uncheck the selected items and check the unselected. To clear all SIP registers at a time, click the **Clear All** button in Figure 3-14.

3.3.4 SIP Account


By default, there is no SIP account available on the gateway. Click **Add New** to add them manually. See Figure 3-16.

Figure 3-16 Add New SIP Account

The table below explains the items shown in above figures.

Item	Description
Index	The unique index of each SIP account.
SIP Trunk No.	The number of the SIP trunk to which the SIP account is registered.
Username	The registration username of the SIP account. Once the SIP account is successfully registered, the SIP server can initiate calls to the gateway via Username .
Password	The registration password of the SIP account. To register the SIP account to the SIP trunk, both configuration items Username and Password should be filled in.
Register Expires	The validity period of the SIP account registry. Once the registry is overdue, the SIP account should be registered again. Range of value: 10~3600, calculated by s, with the default value of 3600.
Register Status	The registration status of the SIP account. It is either <i>Registered</i> or <i>Failed</i> .
Authentication Username	Authentication username of a port, used to register the port to the SIP server when IMS network is enabled. Note: This item appears only when IMS Network is enabled on the SIP trunk corresponding to this SIP account.
Description	More information about each SIP account.

After configuration, click **Save** to save the settings into the gateway or click **Close** to cancel the settings. See Figure 3-17 for the SIP Account Settings Interface

SIP Account								
Check	Index	SIP Trunk No.	Username	Authentication Username	Register Expires (s)	Register Status	Description	Modify
<input type="checkbox"/>	0	0	111		3600	Failed	default	

1 Items Total 20 Items/Page 1/1 First Previous Next Last Go to Page 1 1 Pages Total

Figure 3-17 SIP Account Settings Interface

Click **Modify** in Figure 3-17 to modify a SIP account. See Figure 3-18 for the SIP account modification interface. The configuration items on this interface are the same as those on the **Add New SIP Account** interface.

SIP Account

Index:

SIP Trunk No.:

Username:

Password:

Register Expires (s):

Authentication Username:

Description:

Figure 3-18 Modify SIP Account

To delete a SIP account, check the checkbox before the corresponding index in Figure 3-17 and click the **Delete** button. **Check All** means to select all available items on the current page; **Uncheck All** means to cancel all selections on the current page; **Inverse** means to uncheck the selected items and check the unselected. To clear all SIP accounts at a time, click the **Clear All** button in Figure 3-17.

3.3.5 SIP Trunk Group

By default, there is no SIP trunk group available on the gateway. Click **Add New** to add them manually. See Figure 3-19.

Figure 3-19 Add New SIP Trunk Group

The table below explains the items shown in Figure 3-19.

Item	Description										
Index	The unique index of each SIP trunk group, which is mainly used in the configuration of routing rules and number manipulation rules to correspond to SIP trunk groups.										
Description	More information about each SIP trunk group.										
SIP Trunk Select Mode	<p>When the SIP trunk group receives a call, it will choose a SIP trunk based on the select mode set by this configuration item to ring. The optional values and their corresponding meanings are described in the table below.</p> <table> <tr> <th>Option</th><th>Description</th></tr> <tr> <td><i>Increase</i></td><td>Search for an idle SIP trunk in the ascending order of the SIP trunk number, starting from the minimum.</td></tr> <tr> <td><i>Decrease</i></td><td>Search for an idle SIP trunk in the descending order of the SIP trunk number, starting from the maximum.</td></tr> <tr> <td><i>Cyclic Increase</i></td><td>Provided SIP Trunk N is the available SIP trunk found last time. Search for an idle SIP trunk in the ascending order of the SIP trunk number, starting from SIP Trunk N+1.</td></tr> <tr> <td><i>Cyclic Decrease</i></td><td>Provided SIP Trunk N is the available SIP trunk found last time. Search for an idle SIP trunk in the descending order of the SIP trunk number, starting from SIP Trunk N-1.</td></tr> </table>	Option	Description	<i>Increase</i>	Search for an idle SIP trunk in the ascending order of the SIP trunk number, starting from the minimum.	<i>Decrease</i>	Search for an idle SIP trunk in the descending order of the SIP trunk number, starting from the maximum.	<i>Cyclic Increase</i>	Provided SIP Trunk N is the available SIP trunk found last time. Search for an idle SIP trunk in the ascending order of the SIP trunk number, starting from SIP Trunk N+1.	<i>Cyclic Decrease</i>	Provided SIP Trunk N is the available SIP trunk found last time. Search for an idle SIP trunk in the descending order of the SIP trunk number, starting from SIP Trunk N-1.
Option	Description										
<i>Increase</i>	Search for an idle SIP trunk in the ascending order of the SIP trunk number, starting from the minimum.										
<i>Decrease</i>	Search for an idle SIP trunk in the descending order of the SIP trunk number, starting from the maximum.										
<i>Cyclic Increase</i>	Provided SIP Trunk N is the available SIP trunk found last time. Search for an idle SIP trunk in the ascending order of the SIP trunk number, starting from SIP Trunk N+1.										
<i>Cyclic Decrease</i>	Provided SIP Trunk N is the available SIP trunk found last time. Search for an idle SIP trunk in the descending order of the SIP trunk number, starting from SIP Trunk N-1.										
SIP Trunks	The SIP trunks in the SIP trunk group. If the checkbox before a SIP trunk is grey, it indicates that the SIP trunk has been occupied. The ticked SIP trunks herein will be displayed in the column 'SIP Trunks' in Figure 3-20.										

After configuration, click **Save** to save the settings into the gateway or click **Cancel** to cancel the settings. See Figure 3-20 for the SIP Trunk Group Setting Interface.

SIP Trunk Group					
Check	Index	SIP Trunks	SIP Trunk Select Mode	Description	Modify
<input type="checkbox"/>	0	0	Increase	default	
<input type="checkbox"/>	1	1	Increase	default	
<input type="checkbox"/>	2	2	Increase	default	
<input type="checkbox"/>	3	3	Increase	default	

4 Items Total 20 Items/Page 1/1 First Previous Next Last Go to Page 1 1 Pages Total

Figure 3-20 SIP Trunk Group Settings Interface

Click **Modify** in Figure 3-20 to modify a SIP trunk group. See Figure 3-21 for the SIP trunk group modification interface. The configuration items on this interface are the same as those on the **Add New SIP Trunk Group** interface.

Modify SIP Trunk Group

Index:

Description:

SIP Trunk Select Mode:

SIP Trunks:

☒ 0
☐ 1
☐ 2
☐ 3

Figure 3-21 Modify SIP Trunk Group

To delete a SIP trunk group, check the checkbox before the corresponding index in Figure 3-20 and click the **Delete** button. **Check All** means to select all available items on the current page; **Uncheck All** means to cancel all selections on the current page; **Inverse** means to uncheck the selected items and check the unselected. To clear all SIP trunk groups at a time, click the **Clear All** button in Figure 3-20.

3.3.6 Media Settings

Media Parameters

DTMF Transmit Mode

RFC2833

RFC2833 Payload

101

RTP Port Range

6000,10000

Silence Suppression

Disable

Noise Reduction

Enable

JitterMode

Static Mode

JitterBuffer(ms)

100

JitterUnderrunLead(ms)

100

JitterOverrunLead(ms)

50

Voice Gain Output from IP(dB)

0

CODEC Setting

Priority	CODEC	Packing Time(ms)	Bit Rate (kbps)
1	G711A	20	64
2	G711U	20	64
3	G729	20	8
4	G723	30	6.3
5	G722	30	64
6	AMR	20	12.20
7	iLBC	20	15.2

Save

Reset

Figure 3-22 Media Settings Interface

See Figure 3-22 for the media settings interface where you can configure the RTP port and payload type depending on your requirements. After configuration, click **Save** to save your settings into the gateway or click **Reset** to restore the configurations. If a dialog box pops up after you save your settings asking you to restart the service, do it immediately to apply the changes. Refer to [3.7.16 Restart](#) for detailed instructions. The table below explains the items shown in Figure 3-22.

Item	Description
DTMF Transmit Mode	Sets the mode for the IP channel to send DTMF signals. The optional values are <i>RFC2833</i> , <i>In-band</i> and <i>Signaling</i> , with the default value of <i>RFC2833</i> .
RFC2833 Payload	Payload of the RFC2833 formatted DTMF signals on the IP channel. Range of value: 90~127, with the default value of 101.

RTP Port Range	Supported RTP port range for the IP end to establish a call conversation, with the lower limit of 6000 and the upper limit of 60000 and the difference between larger than 2048. The default value is 6000-10000.																								
Silence Suppression	Sets whether to send comfort noise packets to replace RTP packets or never to send RTP packets to reduce the bandwidth usage when there is no voice signal throughout an IP conversation. The optional values are <i>Enable</i> and <i>Disable</i> , with the default value of <i>Disable</i> . Note: When G723 is selected as CODEC, this configuration setting will turn to Enable automatically.																								
Noise Reduction	Once this feature is enabled, the volume of the noise accompanied with the line will be reduced automatically. The default setting is <i>Enable</i> .																								
JitterMode	Sets the working mode of JitterBuffer. The optional values are <i>Static Mode</i> and <i>Adaptive Mode</i> , with the default value of <i>Static Mode</i> .																								
JitterBuffer	Acceptable jitter for data packets transmission over IP, which indicates the buffering capacity. A larger JitterBuffer means a higher jitter processing capability but as well as an increased voice delay, while a smaller JitterBuffer means a lower jitter processing capability but as well as a decreased voice delay. Range of value: 0~280, calculated by ms, with the default value of 100.																								
JitterUnderrunLead	Sets the initial delay applied to received packets upon accepting packets later than the expected value set in JitterBuffer Item. Range of value: 0~280, calculated by ms, with the default value of 100, Note: Only when JitterMode is set to <i>Static Mode</i> will this item be shown.																								
JitterOverrunLead	Sets the beforehand time inserted if receiving packets is ahead of time (the time of receiving is earlier than 300 minus the value set in JitterBuffer). Rnage of value: 0~280, calculated by ms, with the default value of 50, Note: Only when JitterMode is set to <i>Static Mode</i> will this item be shown.																								
JitterMin	Sets the minimum delay that can be set by the adaptive jitter function. It must be smaller than the value set in JitterBuffer. Range of value: 0~280, calculated by ms, with the default value of 80. Note: Only when JitterMode is set to <i>Adaptive Mode</i> will this item be shown.																								
JitterDecreaseRatio	Sets the rate of delay reduction under the adaptive mode. It defines the maximum percentage of the silence that can be removed in delay reduction. Range of value: 0~100, with the default value of 50, Note: Only when JitterMode is set to <i>Adaptive Mode</i> will this item be shown.																								
JitterIncreaseMax	Sets the maximum delay increased during a silence period. Range of value: 0~280, calculated by ms, with the default value of 30, Note: Only when JitterMode is set to <i>Adaptive Mode</i> will this item be shown.																								
Voice Gain Output from IP	Adjusts the voice gain of call from IP to IP. The value must be a multiple of 3. Range of value: -24~24, calculated by dB, with the default value of 0.																								
CODEC Setting	Sets CODECs for the IP end to establish a call conversation. The table below explains the sub-items: <table border="1"> <thead> <tr> <th>Sub-item</th><th>Description</th></tr> </thead> <tbody> <tr> <td>G711</td><td>Supports G711A and G711E. The default value is G711A.</td></tr> <tr> <td>G723</td><td>Supports G723_1 and G723_2. The default value is G723_1.</td></tr> <tr> <td>G729</td><td>Supports G729B and G729AB. The default value is G729B.</td></tr> <tr> <td>AMR-NB</td><td>Supports AMR-NB_3GPP and AMR-NB_GSM. The default value is AMR-NB_3GPP.</td></tr> <tr> <td>AMR-WB</td><td>Supports AMR-WB_3GPP and AMR-WB_GSM. The default value is AMR-WB_3GPP.</td></tr> <tr> <td>EVRC-B</td><td>Supports EVRC-B_3GPP and EVRC-B_GSM. The default value is EVRC-B_3GPP.</td></tr> <tr> <td>EVRC-WB</td><td>Supports EVRC-WB_3GPP and EVRC-WB_GSM. The default value is EVRC-WB_3GPP.</td></tr> <tr> <td>LD-C</td><td>Supports LD-C_3GPP and LD-C_GSM. The default value is LD-C_3GPP.</td></tr> <tr> <td>LD-WB</td><td>Supports LD-WB_3GPP and LD-WB_GSM. The default value is LD-WB_3GPP.</td></tr> <tr> <td>ISAC</td><td>Supports ISAC_3GPP and ISAC_GSM. The default value is ISAC_3GPP.</td></tr> <tr> <td>ISAC-WB</td><td>Supports ISAC-WB_3GPP and ISAC-WB_GSM. The default value is ISAC-WB_3GPP.</td></tr> </tbody> </table>	Sub-item	Description	G711	Supports G711A and G711E. The default value is G711A.	G723	Supports G723_1 and G723_2. The default value is G723_1.	G729	Supports G729B and G729AB. The default value is G729B.	AMR-NB	Supports AMR-NB_3GPP and AMR-NB_GSM. The default value is AMR-NB_3GPP.	AMR-WB	Supports AMR-WB_3GPP and AMR-WB_GSM. The default value is AMR-WB_3GPP.	EVRC-B	Supports EVRC-B_3GPP and EVRC-B_GSM. The default value is EVRC-B_3GPP.	EVRC-WB	Supports EVRC-WB_3GPP and EVRC-WB_GSM. The default value is EVRC-WB_3GPP.	LD-C	Supports LD-C_3GPP and LD-C_GSM. The default value is LD-C_3GPP.	LD-WB	Supports LD-WB_3GPP and LD-WB_GSM. The default value is LD-WB_3GPP.	ISAC	Supports ISAC_3GPP and ISAC_GSM. The default value is ISAC_3GPP.	ISAC-WB	Supports ISAC-WB_3GPP and ISAC-WB_GSM. The default value is ISAC-WB_3GPP.
Sub-item	Description																								
G711	Supports G711A and G711E. The default value is G711A.																								
G723	Supports G723_1 and G723_2. The default value is G723_1.																								
G729	Supports G729B and G729AB. The default value is G729B.																								
AMR-NB	Supports AMR-NB_3GPP and AMR-NB_GSM. The default value is AMR-NB_3GPP.																								
AMR-WB	Supports AMR-WB_3GPP and AMR-WB_GSM. The default value is AMR-WB_3GPP.																								
EVRC-B	Supports EVRC-B_3GPP and EVRC-B_GSM. The default value is EVRC-B_3GPP.																								
EVRC-WB	Supports EVRC-WB_3GPP and EVRC-WB_GSM. The default value is EVRC-WB_3GPP.																								
LD-C	Supports LD-C_3GPP and LD-C_GSM. The default value is LD-C_3GPP.																								
LD-WB	Supports LD-WB_3GPP and LD-WB_GSM. The default value is LD-WB_3GPP.																								
ISAC	Supports ISAC_3GPP and ISAC_GSM. The default value is ISAC_3GPP.																								
ISAC-WB	Supports ISAC-WB_3GPP and ISAC-WB_GSM. The default value is ISAC-WB_3GPP.																								

<i>Priority</i>	Priority for choosing the CODEC in an SIP conversation. The smaller the value is, the higher the priority will be.
<i>CODEC</i>	Seven optional CODECs are supported: <i>G711A</i> , <i>G711U</i> , <i>G729</i> , <i>G723</i> , <i>G722</i> , <i>AMR</i> and <i>iLBC</i> .
<i>Packing Time</i>	Time interval for packing an RTP packet, calculated by ms.
<i>Bit Rate</i>	The number of thousand bits (excluding the packet header) that are conveyed per second.

By default, all of the seven CODECs are supported and ordered *G711A*, *G711U*, *G729*, *G723*, *G722*, *AMR* and *iLBC* by priority from high to low. The CODECs set here will be the default CODEC for the new added SIP trunks.

The packing time and bit rate supported by different CODECs are listed in the table below. Those values in bold face are the default values.

COEDC	Packing Time (ms)	Bit Rate (kbps)
<i>G711A</i>	5 / 10 / 20 / 30 / 40 / 50 / 60	64
<i>G711U</i>	5 / 10 / 20 / 30 / 40 / 50 / 60	64
<i>G729</i>	20	8
<i>G723</i>	30 / 60 / 90	5.3 / 6.3
<i>G722</i>	5 / 10 / 20 / 30 / 40	64
<i>AMR</i>	20 / 40 / 60 / 80 / 100	4.75 / 5.15 / 5.90 / 6.70 / 7.40 / 7.95 / 10.20 / 12.20
<i>iLBC</i>	20 / 30 / 40 / 60	15.2

3.4 Route Settings

Route Settings is used to specify the routing rules for calls from IP→IP. See Figure 3-23.

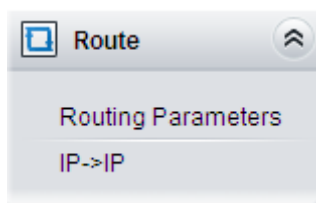


Figure 3-23 Route Settings

3.4.1 Routing Parameters

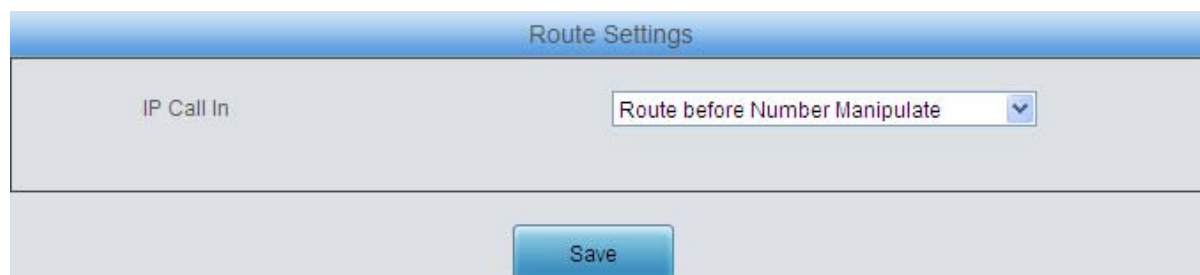


Figure 3-24 Routing Parameters Configuration Interface

See Figure 3-24 for the routing parameters configuration interface. On this interface, you can set the routing rules for calls from IP→IP to be routing before or after number manipulation. The default value is *Route before Number Manipulate*.

After configuration, click **Save** to save the above settings into the gateway.

3.4.2 IP to IP



By default, there is no IP→IP routing rule available on the gateway. Click **Add New** to add them manually. See Figure 3-25 for the IP→IP routing rule adding interface.

Figure 3-25 Add New Routing Rule (IP→IP)

The table below explains the items shown in the above figure.

Item	Description
Index	The unique index of each routing rule, which denotes its priority. A routing rule with a smaller index value has a higher priority. If a call matches several routing rules, it will be processed according to the one with the highest priority.
Call Initiator	SIP trunk group from where the call is initiated. This item can be set to a specific SIP trunk group or SIP Trunk Group [ANY] which indicates any SIP trunk group.
CallerID Prefix, Calleed Prefix	A string of numbers at the beginning of the calling/called party number. This item can be set to a specific string or "*" which indicates any string. These two configuration items together with Call Initiator can specify the calls which apply to a routing rule. Note: Multiple rules are supported for CallerID/Calleed prefix. They are separated by ":".
Call Destination	SIP trunk group to which the call will be routed.
Number Filter	Number filter rule which will be applicable to this route. It is set in Number Filter . See 3.5.4 Filtering Rule for details.
Description	More information about each routing rule.

After configuration, click **Save** to save the settings into the gateway or click **Close** to cancel the settings. See Figure 3-26 for the IP→IP Routing Rule Configuration Interface.

Routing Rules								
Check	Index	Call Initiator	CallerID Prefix	CalleeID Prefix	Number Filter	Call Destination	Description	Modify
<input type="checkbox"/>	63	SIP Trunk Group [0]	*	*	none	SIP Trunk Group [1]	default	
<input type="checkbox"/>	62	SIP Trunk Group [3]	*	*	none	SIP Trunk Group [2]	default	

2 Items Total 20 Items/Page 1/1 First Previous Next Last Go to Page 1 1 Pages Total

Figure 3-26 IP→IP Routing Rule Configuration Interface

Click **Modify** in Figure 3-26 to modify a routing rule. See Figure 3-27 for the IP→IP routing rule modification interface. The configuration items on this interface are the same as those on the **Add New Routing Rule (IP→IP)** interface. Note that the item **Index** cannot be modified.

IP→PSTN Routing Rule

Index: 63

Call Initiator: SIP Trunk Group [0]

CallerId Prefix: 333[1,3]:444[6,9]

CalleeID Prefix: *

Call Destination: PCM Trunk Group [0]

Number Filter: none

Description: default

Figure 3-27 Modify Routing Rule (IP→IP)

To delete a routing rule, check the checkbox before the corresponding index in Figure 3-26 and click the **Delete** button. **Check All** means to select all available items on the current page; **Uncheck All** means to cancel all selections on the current page; **Inverse** means to uncheck the selected items and check the unselected. To clear all routing rules at a time, click the **Clear All** button in Figure 3-26.

3.5 Number Filter

Number Filter includes four parts: **Whitelist**, **Blacklist**, **Number Pool** and **Filtering Rule**. See Figure 3-28.



Figure 3-28 Number Filter Interface

3.5.1 Whitelist

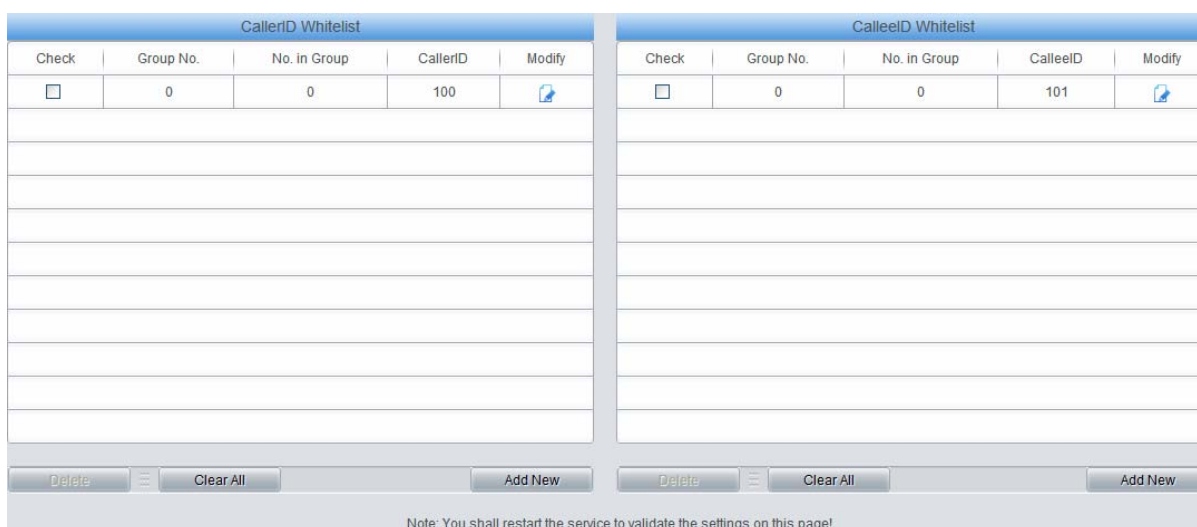


Figure 3-29 Whitelist Setting Interface

See Figure 3-29 for the Whitelist Setting Interface, which includes two parts: **CallerID Whitelist** and **CalleeID Whitelist**.

A new CallerID/CalleeID whitelist can be added by the **Add New** button. See Figure 3-30, Figure 3-31 for CallerID/CalleeID whitelist adding interface.

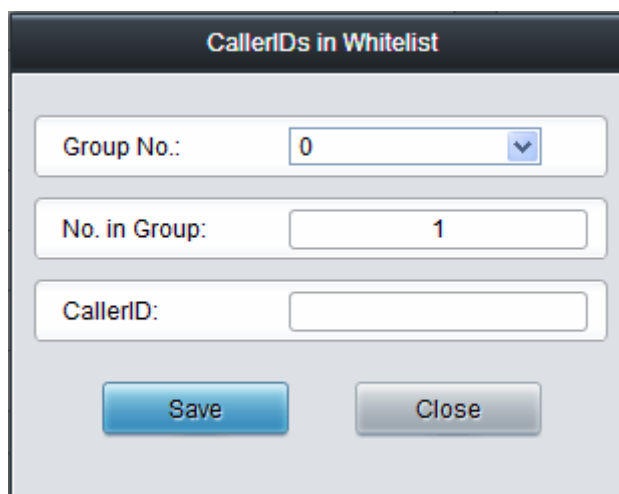


Figure 3-30 Add New CallerIDs in Whitelist Interface



Figure 3-31 Add New CalleeIDs in Whitelist Interface

The table below explains the items shown in above figures.

Item	Description
Group	The corresponding Group ID for CallerIDs/CalleeIDs in the whitelist. The value range is 0~7.
No. in Group	The corresponding No. for different CallerIDs/CalleeIDs in a same group.
CallerID	CallerID in the whitelist, which must be filled in with numbers or "*" (indicating any string) and can not be left empty. Example: 135*1 denotes any CallerIDs which start from 135 and end with 1 will be accepted.
CalleeID	CalleeID in the whitelist, which must be filled in with numbers or "*" (indicating any string) and can not be left empty. Example: 135*1 denotes any CalleeIDs which start from 135 and end with 1 will be accepted.

After configuration, click **Save** to save the above settings into the gateway or click **Close** to cancel the settings.

Click **Modify** in Figure 3-29 to modify the CallerID or CalleeID whitelist. See Figure 3-32, Figure 3-33 for CallerIDs/CalleeIDs on the Whitelist Modification interface. The configuration items on this interface are the same as those on the **Add New CallerIDs/CalleeIDs in Whitelist** interface. The item *Group No.* cannot be modified.



Figure 3-32 Modify CallerIDs in Whitelist

CalleelIDs in Whitelist

Group:

0

No. in Group:

0

CalleelID:

101

Save

Close

Figure 3-33 Modify CalleeIDs in Whitelist

To delete a CallerIDs/CalleeIDs in the whitelist, check the checkbox before the corresponding index in Figure 3-29 and click the '**Delete**' button. To clear all CallerIDs/CalleeIDs in the whitelist at a time, click the **Clear All** button in Figure 3-29.

Note: If a CallerID or CalleeID set in the whitelist is the same as one in the blacklist, it will go invalid. That is, the blacklist has a higher priority than the whitelist. The total amount of numbers in both whitelist and blacklist cannot exceed 5000.

3.5.2 Blacklist

CallerID Blacklist

Check	Group No.	No. in Group	CallerID	Modify
<input type="checkbox"/>	0	0	78	
<input type="checkbox"/>	0	1	111	

Delete
Clear All
Add New

CalleeID Blacklist

Check	Group No.	No. in Group	CalleeID	Modify
<input type="checkbox"/>	0	0	111	

Delete
Clear All
Add New

Note: You shall restart the service to validate the settings on this panel.

Figure 3-34 Blacklist Setting Interface

The Blacklist Setting interface is almost the same as the Whitelist Setting interface; only the whitelist changes to the blacklist. See Figure 3-34. The configuration items on this interface are the same as those on the Whitelist Setting interface (Figure 3-30, Figure 3-31).

Note: The lacklist has a higher priority than the whitelist. That is, if the same number exists in both blacklist and whitelist, the number in blacklist has priority.

3.5.3 Number Pool










Number Pool					
Check	Group No.	No. in Group	Initiative Number	Number Amount	Modify
<input type="checkbox"/>	0	0	10000	2	
<input type="checkbox"/>	0	1	1000	1	
<input type="checkbox"/>	0	2	1000	1	
<input type="checkbox"/>	0	3	1000	1	
<input type="checkbox"/>	0	4	1000	1	
<input type="checkbox"/>	0	5	200	1	
<input type="checkbox"/>	0	6	1000	1	
<input type="checkbox"/>	1	0	200	1	
<input type="checkbox"/>	2	0	111	2	
<input type="checkbox"/>	3	0	222	2	
<input type="checkbox"/>	4	0	333	2	

Figure 3-35 Number Pool Setting Interface

See Figure 3-35 for the Number Pool Setting interface. A new number pool can be added by the **Add New** button on the bottom right corner of the list in the above figure. See Figure 3-36 for the Number Pool adding interface.

Number Pool

Group:

0

No. in Group:

7

Starting Number:

Number Amount

Save

Close

Figure 3-36 Add New Number Pool

The table below explains the items shown in the above figure.

Item	Description
Group	The corresponding Group ID for numbers in the number pool. The value range is 0~15.
No. in Group	The corresponding No. for different numbers in a same group. It supports up to 100 numbers in one group.
Starting Number	The starting number in a number Pool. It must be filled in with numbers and can not be left empty.
Number Amount	The amount of the numbers in the number pool. The value range is 1~999999999.

After configuration, click **Save** to save the above settings into the gateway or click **Close** to cancel the settings.

Click **Modify** in Figure 3-35 to modify the number pool. See Figure 3-37 for the number pool modification interface. The configuration items on this interface are the same as those on the **Add New Number Pool** interface.

Figure 3-37 Modify Number Pool Interface

To delete a number pool, check the checkbox before the corresponding index in Figure 3-35 and click the '**Delete**' button. To clear all number pools at a time, click the **Clear All** button in Figure 3-35.

3.5.4 Filtering Rule

Filtering Rule										
Check	No.	CallerID Whitelist	CalleeID Whitelist	CallerID Blacklist	CalleeID Blacklist	CallerID Pool in Whitelist	CallerID Pool in Blacklist	CalleeID Pool in Whitelist	CalleeID Pool in Blacklist	Original Calle
<input type="checkbox"/>	0	0	none	none	none	0	none	none	none	
<input type="checkbox"/>	1	none	none	none	none	none	none	none	none	
<input type="checkbox"/>	2	none	none	none	none	none	none	none	none	
<input type="checkbox"/>	3	none	none	none	none	none	none	none	none	
<input type="checkbox"/>	4	none	none	none	none	none	none	none	none	
<input type="checkbox"/>	5	none	none	none	none	none	none	none	none	
<input type="checkbox"/>	6	none	none	none	none	none	none	none	none	
<input type="checkbox"/>	7	none	none	none	none	none	none	none	none	
<input type="checkbox"/>	8	none	none	none	none	none	none	none	none	
<input type="checkbox"/>	9	none	none	none	none	none	none	none	none	
<input type="checkbox"/>	10	none	none	none	none	none	none	none	none	
<input type="checkbox"/>	11	none	none	none	none	none	none	none	none	

12 Items Total 15 Items/Page 1/1 First Previous Next Last Go to Page 1 1 Pages Total

Figure 3-38 Filtering Rule Setting Interface

See Figure 3-38 for the Filtering Rule Setting Interface. A new filtering rule can be added by the **Add New** button on the bottom right corner of the list in the above figure. See Figure 3-39 for the Filtering Rule Adding interface.

Filtering Rule

No.:

CallerID Whitelist:

none

CalleeID Whitelist:

none

CallerID Blacklist:

none

CalleeID Blacklist:

none

CallerID Pool in Whitelist:

none

CallerID Pool in Blacklist:

none

CalleeID Pool in Whitelist:

none

CalleeID Pool in Blacklist:

none

Original CalleeID Pool in Whitelist:

none

Original CalleeID Pool in Blacklist:

none

Description:

Save

Close

Figure 3-39 Add New Filtering Rule

The table below explains the items shown in the above figure.

Item	Description
No.	The corresponding number for a filtering rule. The value range is 0~99.
CallerID Whitelist	The Group No. of CallerIDs saved on the whitelist setting interface.
CalleeID Whitelist	The Group No. of CalleeIDs saved on the whitelist setting interface.
CallerID Blacklist	The Group No. of CallerIDs saved on the blacklist setting interface.
CalleeID Blacklist	The Group No. of CalleeIDs saved on the blacklist setting interface.
CallerID Pool in Whitelist	Select a Group No. which is set in the whitelist from the number pool as the CallerID pool in whitelist.
CallerID Pool in Blacklist	Select a Group No. which is set in the blacklist from the number pool as the CallerID pool in blacklist.
CalleeID Pool in Whitelist	Select a Group No. which is set in the whitelist from the number pool as the CalleeID pool in whitelist.

CalleeID Pool in Blacklist	Select a Group No. which is set in the blacklist from the number pool as the CalleeID pool in blacklist.
Original CalleeID Pool in Whitelist	Select a Group No. which is set in the whitelist from the number pool as the original CalleeID pool in whitelist.
Original CalleeID Pool in Blacklist	Select a Group No. which is set in the blacklist from the number pool as the original CalleeID pool in blacklist.
Description	Remarks for the filtering rule. It can be any information, but can not be left empty.

After configuration, click **Save** to save the above settings into the gateway or click **Close** to cancel the settings.

Click **Modify** in Figure 3-38 to modify the filtering rule. See Figure 3-40 for the filtering rule modification interface. The configuration items on this interface are the same as those on the **Add New Filtering Rule** interface.

The 'Filtering Rule' interface includes the following configuration items:

- No.:** 0
- CallerID Whitelist:** 0
- CalleeID Whitelist:** none
- CallerID Blacklist:** none
- CalleeID Blacklist:** none
- CallerID Pool in Whitelist:** 0
- CallerID Pool in Blacklist:** none
- CalleeID Pool in Whitelist:** none
- CalleeID Pool in Blacklist:** none
- Original CalleeID Pool in Whitelist:** 0
- Original CalleeID Pool in Blacklist:** none
- Description:** default

Buttons: Save, Close

Figure 3-40 Modify Filtering Rule Interface

To delete a filtering rule, check the checkbox before the corresponding index in Figure 3-38 and

click the '**Delete**' button. To clear all filtering rules at a time, click the **Clear All** button in Figure 3-38.

3.6 Number Manipulation

Number Manipulation includes two parts: **IP→IP CallerID** and **IP→IP CalleeID**. See Figure 3-41.

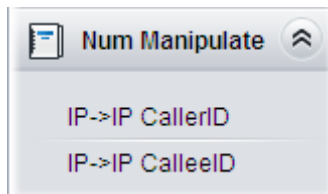


Figure 3-41 Number Manipulation

3.6.1 IP to IP CallerID

By default, there is no IP→IP CallerID manipulation available on the gateway. Click **Add New** to add them manually. See Figure 3-42 for the IP→IP CallerID manipulation rule adding interface.

A screenshot of the 'IP->IP CallerID Manipulation' configuration window. It features a dark header with the title. Below the header are several input fields and dropdown menus: 'Index' (63), 'Call Initiator' (SIP Trunk Group [0]), 'CallerID Prefix' (*), 'CalleeID Prefix' (*), 'With Original CalleeID' (No), 'Stripped Digits from Left' (0), 'Stripped Digits from Right' (0), 'Reserved Digits from Right' (0), 'Prefix to Add' (empty), 'Suffix to Add' (empty), and 'Description' (default). At the bottom are 'Save' and 'Close' buttons.

Figure 3-42 Add IP→IP CallerID Manipulation Rule

The table below explains the items shown in the above figure.

Item	Description
Index	The unique index of each number manipulation rule, which denotes its priority. A number manipulation rule with a smaller index value has a higher priority. If a call matches several number manipulation rules, it will be processed according to the one with the highest priority.
Call Initiator	SIP trunk group from where the call is initiated. This item can be set to a specific SIP trunk group or SIP Trunk Group[ANY] which indicates any SIP trunk group.
CallerID Prefix, CalleeID Prefix	A string of numbers at the beginning of the calling/called party number. This item can be set to a specific string or "*" which indicates any string. These two configuration items together with Call Initiator and With Original CalleeID can specify the calls which apply to a number manipulation rule.
With Original CalleeID	If this item is set to Yes, it indicates that the number manipulation rule is only applicable to the calls with original CalleeID/redirecting number. The default value is <i>No</i> .
Stripped Digits from Left	The amount of digits to be deleted from the left end of the number. If the value of this item exceeds the length of the current number, the whole number will be deleted.
Stripped Digits from Right	The amount of digits to be deleted from the right end of the number. If the value of this item exceeds the length of the current number, the whole number will be deleted.
Reserved Digits from Right	The amount of digits to be reserved from the right end of the number. Only when the value of this item is less than the length of the current number will some digits be deleted from left; otherwise, the number will not be manipulated.
Prefix to Add	Designated information to be added to the left end of the current number.
Suffix to Add	Designated information to be added to the right end of the current number.
Description	More information about each number manipulation rule.

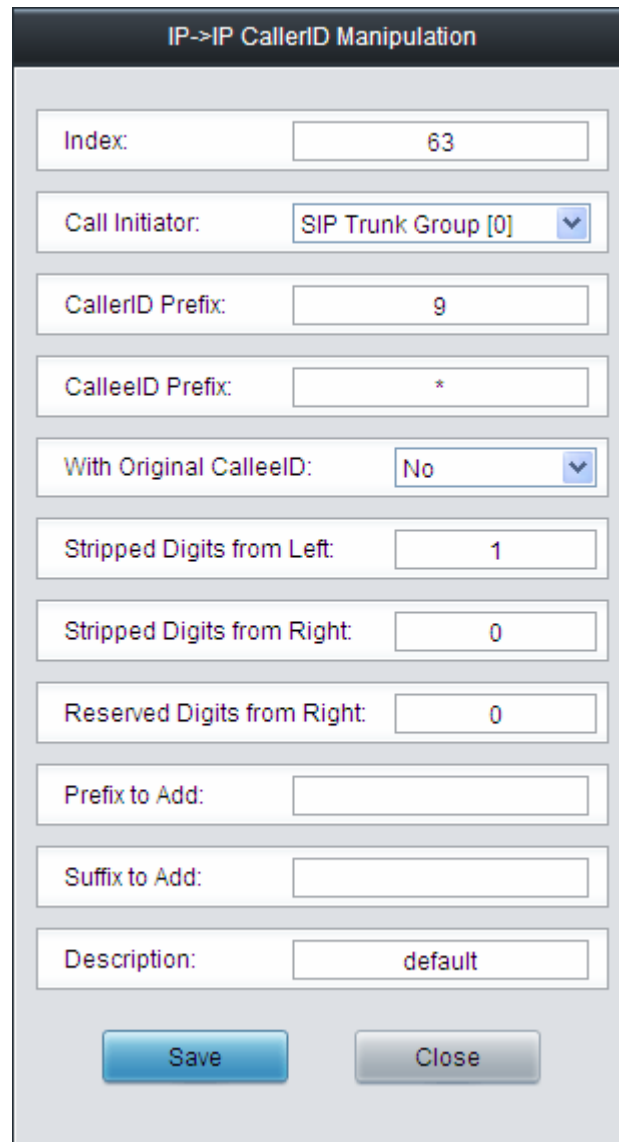
Note: The number manipulation is performed in 5 steps by the order of the following configuration items: **Stripped Digits from Left**, **Stripped Digits from Right**, **Reserved Digits from Right**, **Prefix to Add** and **Suffix to Add**.

After configuration, click **Save** to save the settings into the gateway or click **Close** to cancel the settings. See Figure 3-43 for IP→IP CallerID Manipulation Interface.

Number Manipulation Rules											
Check	Index	Call Initiator	CallerID Prefix	CalleeID Prefix	With Original CalleeID	Stripped Digits from Left	Stripped Digits from Right	Reserved Digits from Right	Prefix to Add	Suffix to Add	Description
<input type="checkbox"/>	83	SIP Trunk Group [0]	9	*	No	1	0	0			default
Check All Uncheck All Inverse Delete Clear All Add New											
1 Items Total 20 Items/Page 1/1 First Previous Next Last Go to Page 1 1 Pages Total											

Figure 3-43 IP→IP CallerID Manipulation Interface

Click **Modify** in Figure 3-43 to modify a number manipulation rule. See Figure 3-44 for the IP→IP CallerID manipulation rule modification interface. The configuration items on this interface are the same as those on the **Add IP→IP CallerID Manipulation Rule** interface. Note that the item **Index** cannot be modified.



The image shows a configuration window titled "IP->IP CallerID Manipulation". It contains several input fields and dropdown menus for configuring a manipulation rule. The fields are as follows:

- Index:** 63
- Call Initiator:** SIP Trunk Group [0]
- CallerID Prefix:** 9
- CalleeID Prefix:** *
- With Original CalleeID:** No
- Stripped Digits from Left:** 1
- Stripped Digits from Right:** 0
- Reserved Digits from Right:** 0
- Prefix to Add:** (empty)
- Suffix to Add:** (empty)
- Description:** default

At the bottom of the window are two buttons: "Save" and "Close".

Figure 3-44 Modify IP→IP CallerID Manipulation Rule

To delete a number manipulation rule, check the checkbox before the corresponding index in Figure 3-43 and click the **Delete** button. **Check All** means to select all available items on the current page; **Uncheck All** means to cancel all selections on the current page; **Inverse** means to uncheck the selected items and check the unselected. To clear all number manipulation rules at a time, click the **Clear All** button in Figure 3-43.

3.6.2 IP to IP CalleeID

The number manipulation process for IP→IP CalleeID is almost the same as that for IP→IP CallerID; only the number to be manipulated changes from CallerID to CalleeID. See Figure 3-45 for IP→IP CalleeID manipulation interface. The configuration items on this interface are the same as those on **IP→IP CallerID Manipulation Interface** (Figure 3-43).


Number Manipulation Rules												
Check	Index	Call Initiator	CallerID Prefix	CalleeID Prefix	With Original CalleeID	Stripped Digits from Left	Stripped Digits from Right	Reserved Digits from Right	Prefix to Add	Suffix to Add	Description	Modify
<input type="checkbox"/>	63	SIP Trunk Group [0]	*	*	No	0	0	0			default	
Check All Uncheck All Inverse Delete Clear All Add New												
1 Items Total 20 Items/Page 1/1 First Previous Next Last Go to Page 1 1 Pages Total												

Figure 3-45 IP→IP CalleeID Manipulation Interface

3.7 System Tools

System Tools is mainly for gateway maintenance. It provides such features as IP modification, time synchronization, data backup, log inquiry and connectivity check. See Figure 3-46 for details.

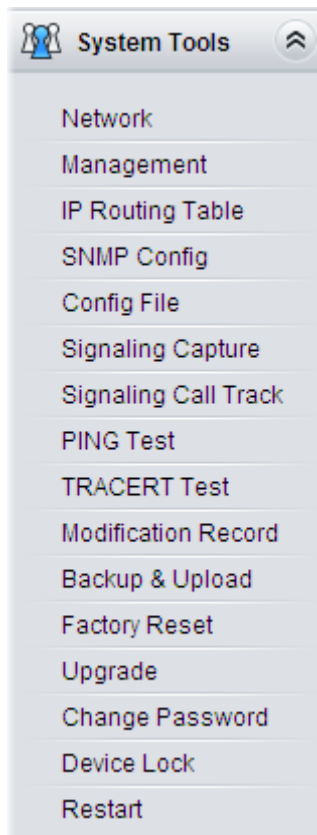


Figure 3-46 System Tools

3.7.1 Network

The screenshot displays the 'Network Settings' window. It is divided into three main sections: LAN 1, LAN 2, and BOND Setting.

LAN 1 Settings:

- IP Address (I): 201.123.112.17
- Subnet Mask (U): 255.255.255.0
- Default Gateway (D): 201.123.112.254
- DNS Server (P): 0.0.0.0
- Speed and Duplex Mode: Automatic Detection (dropdown menu)

LAN 2 Settings:

- IP Address (I): 192.168.0.101
- Subnet Mask (U): 255.255.255.0
- Default Gateway (D): 192.168.0.254
- DNS Server (P): 0.0.0.0
- Speed and Duplex Mode: Automatic Detection (dropdown menu)

BOND Setting:

- BOND: ☒ Yes ☐ No
- BOND Address: LAN 1 (dropdown menu)

At the bottom of the window, there are two buttons: 'Save' and 'Reset'.

Figure 3-47 Network Settings Interface

See Figure 3-47 for the Network Settings interface. A gateway has two network ports, each of which can be configured with independent IP address, subnet mask, default gateway and DNS server. The Bond feature when enabled will make the information of LAN1 and LAN2 duplicated and backed up so as to realize the hot-backup function for LAN1 and LAN2. By default, this feature is *disabled*.

Note: 1. The two configuration items IP Address and Default Gateway cannot be the same for NET 1 and NET 2.

2. By default, *Speed and Duplex Mode* is hidden, set to Automatic Detection. You can click 'F' to let it display. We suggest you do not modify it because the non-automatic detection may cause abnormality in network interface.

If the Network Detect feature is enabled, a ping test will automatically be initiated from this IP address to the gateway to check the connection status between them. By default, this feature is

disabled.

After configuration, click **Save** to save the above settings into the gateway or click **Reset** to restore the configurations. After changing the IP address, you shall log in the gateway again using your new IP address.

3.7.2 Management

The screenshot shows the 'Management Parameters' configuration interface. It is organized into several sections:

- WEB Management:** Includes 'WEB Port' (1080), 'Access Setting' (IPs in Whitelist), and 'IP Address' (201.123.115, 201.123.113). A note states 'IP addresses are separated by ;'.
- SSH Management Config:** Includes 'SSH' (Yes/No radio buttons, with 'Yes' selected) and 'SSH Port' (22).
- Remote Data Capture Config:** Includes 'Remote Data Capture' (Yes/No radio buttons, with 'No' selected).
- SYSLOG Parameters:** Includes 'SYSLOG' (Yes/No radio buttons, with 'Yes' selected), 'Server Address' (127.0.0.1), and 'SYSLOG Level' (ERROR).
- Time Parameters:** Includes 'NTP' (Yes/No radio buttons, with 'Yes' selected), 'NTP Server Address' (127.0.0.1), 'Synchronizing Cycle' (3600 s), 'Daily Restart' (Yes/No radio buttons, with 'Yes' selected), 'Restart Time' (7 h 13 m), 'System Time' (Modify 2014-10-14 16:00:25), and 'Time Zone' (GMT+8:00 (Beijing, Singapore, Taipei, Kuala Lumpur)).

At the bottom of the interface are two buttons: 'Save' and 'Reset'.

Figure 3-48 Management Parameters Setting Interface

See Figure 3-48 for the Management Parameters Setting interface. The table below explains the items shown in the above figure.

Item	Description
WEB Port	The port which is used to access the gateway via WEB. The default value is 80.
Access Setting	Sets the IP addresses which can access the gateway via WEB. By default, all IPs are allowed. You can set an IP whitelist to allow all IPs within it to access the gateway freely. Also can set an IP blacklist to forbid all IPs within it to access the gateway.

SSH	Sets whether to enable the gateway to be accessed via SSH, with the default value of <i>No</i> .
SSH Port	The port which is used to access the gateway via SSH.
Remote Data Capture	After this feature is enabled, you can obtain the gateway data via a remote capture tool, with the default value of <i>No</i> .
SYSLOG	Sets whether to enable SYSLOG. It is required to fill in SYSLOG Server Address and SYSLOG Level in case SYSLOG is enabled. By default, SYSLOG is disabled.
Server Address	Sets the SYSLOG server address for log reception.
SYSLOG Level	Sets the SYSLOG level. There are three options: <i>ERROR</i> , <i>WARNING</i> and <i>INFO</i> .
NTP	Sets whether to enable the NTP time synchronization feature. It is required to fill in NTP Server Address , Synchronizing Cycle and Time Zone in case NTP is enabled. By default, NTP is disabled.
NTP Server Address	Sets the Server address for NTP time synchronization.
Synchronizing Cycle	Sets the cycle for NTP time synchronization.
Daily Restart	Sets whether to restart the gateway regularly every day at the preset Restart Time . By default, this feature is disabled.
Restart Time	Sets the time to restart the gateway regularly.
System Time	The system time. Check the checkbox before Modify and change the time in the edit box.
Time Zone	The time zone of the gateway.

3.7.3 IP Routing Table

IP Routing Table is used to set the route for the LAN port when these two network ports both transport SIP. Thus, the LAN can access some IPs in other different network segment. By default, there is no routing table available on the gateway, click **Add New** to add them manually. See Figure 3-49.

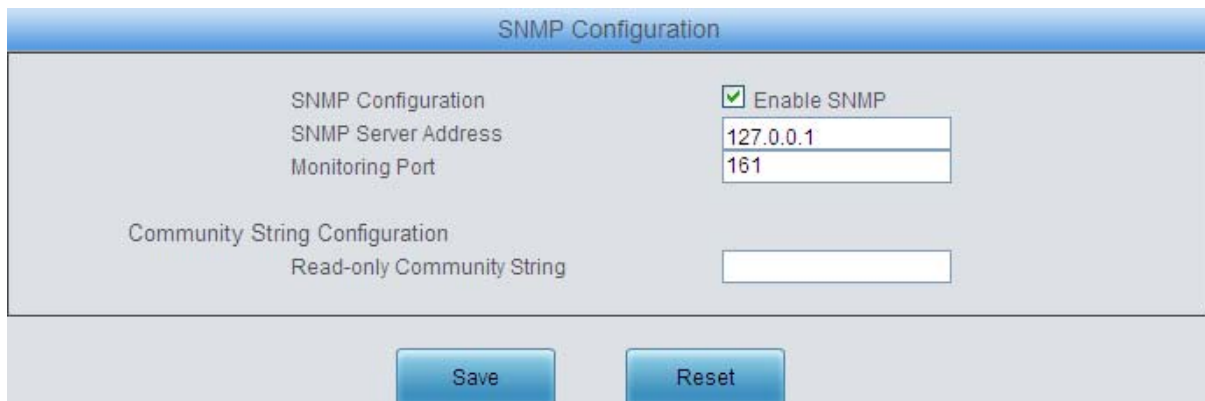
The image shows a 'Routing Table' configuration window. It has a title bar 'Routing Table'. Inside, there are four input fields: 'No.' with the value '0', 'Destination', 'Subnet Mask', and 'Network Port' with a dropdown menu showing 'NET 1'. At the bottom, there are two buttons: 'Save' and 'Close'.

Figure 3-49 Routing Table Adding Interface

The table below explains the items shown in above figures.

Item	Description
------	-------------

3.7.4 SNMP Config



The image shows a web-based configuration interface for SNMP. It has a blue header bar with the text "SNMP Configuration". Below the header, there are two main sections. The first section, "SNMP Configuration", contains a checkbox labeled "Enable SNMP" which is checked, a text input field for "SNMP Server Address" with the value "127.0.0.1", and a text input field for "Monitoring Port" with the value "161". The second section, "Community String Configuration", contains a text input field for "Read-only Community String" which is currently empty. At the bottom of the interface, there are two blue buttons labeled "Save" and "Reset".

Figure 3-52 SNMP Configuration Interface

See Figure 3-52 for the SNMP configuration interface. If the SNMP feature is enabled, once the gateway receives a request from the SNMP management software, it will collect relevant information and reply to the SNMP management software. By default, the SNMP feature is disabled. The available information includes kernel version, CPU usage, processes, memory usage, startup information, NET status and etc. Currently, the gateway only provides the community string for information acquisition.

The table below explains the configuration items shown in the above figure.

Item	Description
SNMP Server Address	IP address of SNMP.
Monitoring Port	Monitoring Port for SNMP on the gateway.
Read-only Community String	Community string used for information acquisition.

You can query OID (object identification trees) = .1.3.6.1.4.1.2021.51 at the SNMP Client to obtain the signaling link status and the line synchronization information,

3.7.5 Configuration File

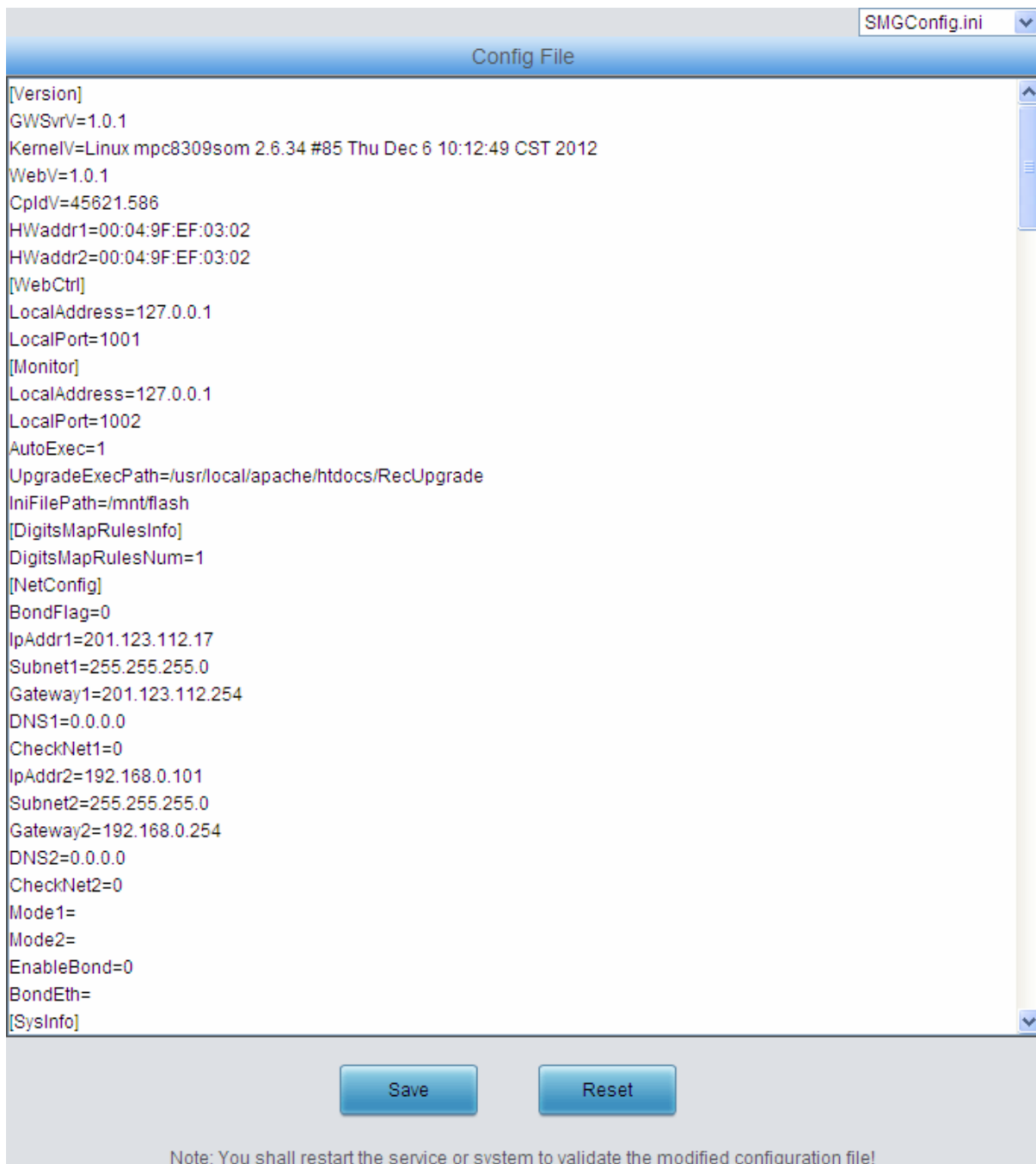


Figure 3-53 Configuration File Interface

See Figure 3-53 for the Configuration File interface, including two files: SMGConfig.ini and ShConfig.ini. You can check and modify the items in these configuration files through this interface. Configurations about the gateway server, such as route rules, number manipulation, number filter and so on, are included in SMGConfig.ini; Configurations about the board are included in ShConfig.ini. You can modify these configurations on the interface directly, and then click **Save** to save the above settings into the gateway or click **Reset** to restore the configurations.

3.7.6 Signaling Capture

The screenshot displays the 'Signaling Capture' interface, which is divided into two main sections: 'Data Capture' and 'IP Two-way Recording'.

Data Capture Section:

- Choose a network interface to capture data:** A dropdown menu is set to 'LAN 1(201.123.112.199)'. To its right is a 'Start' button.
- Capture RTP:** An unchecked checkbox.
- Destination Address for Syslog:** A text input field containing '201.123.112.254'. To its right is a 'Stop' button.

IP Two-way Recording Section:

- Choose a channel group and channel to record data (Row 1):** The first dropdown is 'Channel Group1' and the second is 'Channel 0'. To their right are 'Start' and 'Stop' buttons.
- Choose a channel group and channel to record data (Row 2):** The first dropdown is 'Channel Group1' and the second is 'Channel 16'. To their right are 'Start' and 'Stop' buttons.

Bottom Section:

- Two buttons: 'Clean Data' and 'Download Log'.

Figure 3-54 Signaling Capture Interface

See Figure 3-54 for the Signaling Capture interface. Data Capture is used to capture data on the network interface you choose. Click **Start** to start capturing data (1024000 packets at most) on the corresponding network interface. SIP and SysLog are supported at present. You can input a destination address for syslog to which the syslog file will be sent. Click **Stop** to stop data capture and download the captured packets.

IP Two-way Recording is used to make recording of a designated channel in a specified channel group. Click **Start** to start recording data (maximum consecutively recording time: 1 minute). Click **Stop** to stop data recording and download the recorded data.

Click **Clean Data** to clean all the recording files and captured packages. Click **Download Log** to download such logs as core files, configuration files, error information and so on.

3.7.7 Signaling Call Track

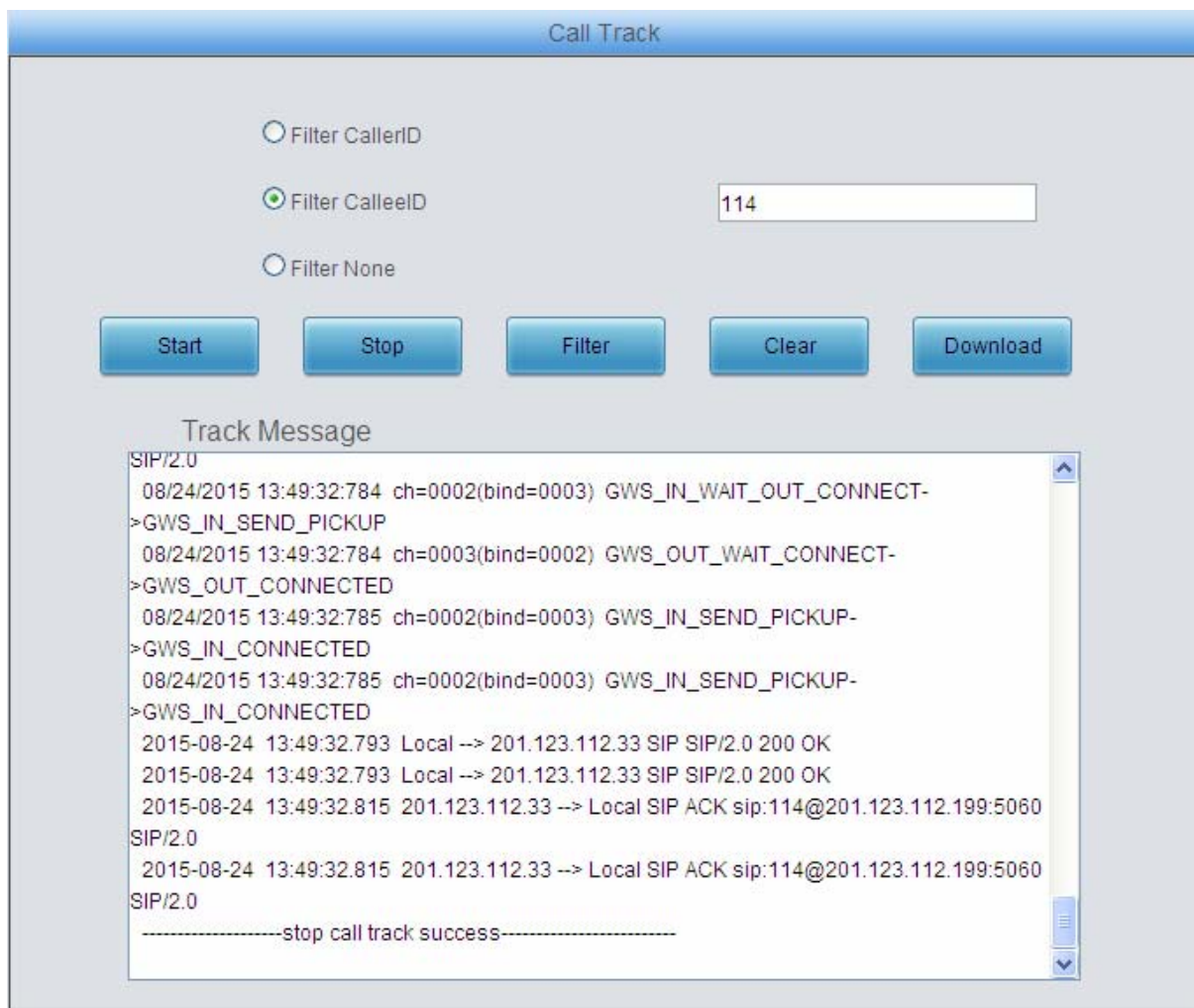


Figure 3-55 Call Track Interface

See Figure 3-55 for the Call Track Interface, including three modes: Filter CallerID, Filter CalleeID and Filter None. This is mainly used to output and save call information, facilitating call trace and problem debugging. Click **Start** to track calls, and the trace logs will be shown in the "Track Message" field; click **Stop** to stop the call track; click **Filter** to filter the trace logs according to the condition you set; click **Clear** to clear all trace logs; click **download** to download trace logs.

3.7.8 PING Test

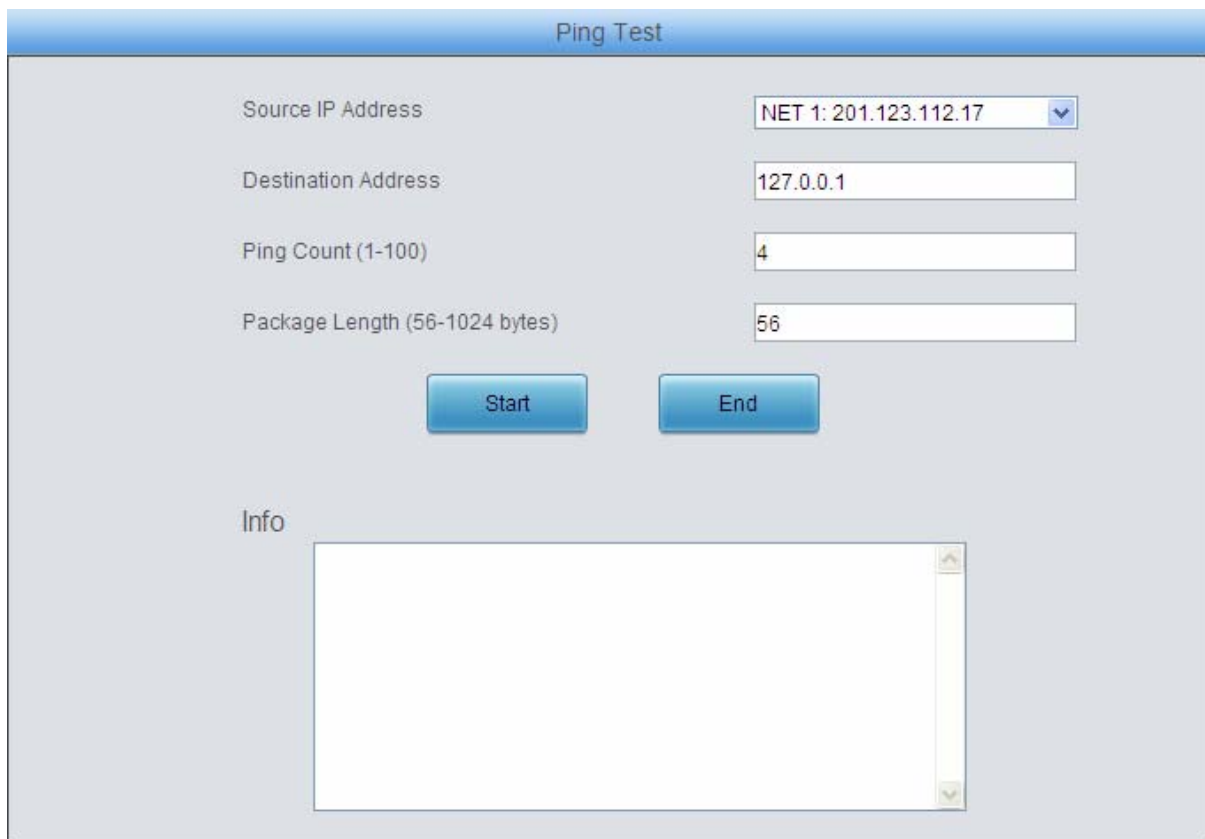


Figure 3-56 Ping Test Interface

See Figure 3-56 for the Ping Test interface. A Ping test can be initiated from the gateway on a designated IP address to check the connection status between them. The table below explains the configuration items shown in the above figure.

Item	Description
Source IP Address	Source IP address where the Ping test is initiated.
Destination Address	Destination IP address on which the Ping test is executed.
Ping Count	The number of times that the Ping test should be executed. Range of value: 1~100.
Package Length	Length of a data package used in the Ping test. Range of value: 56~1024 bytes.
Info	The information returned during the Ping test, helping you to learn the network connection status between the gateway and the destination address.

After configuration, click **Start** to execute the Ping test; click **End** to terminate it immediately.

3.7.9 TRACERT Test

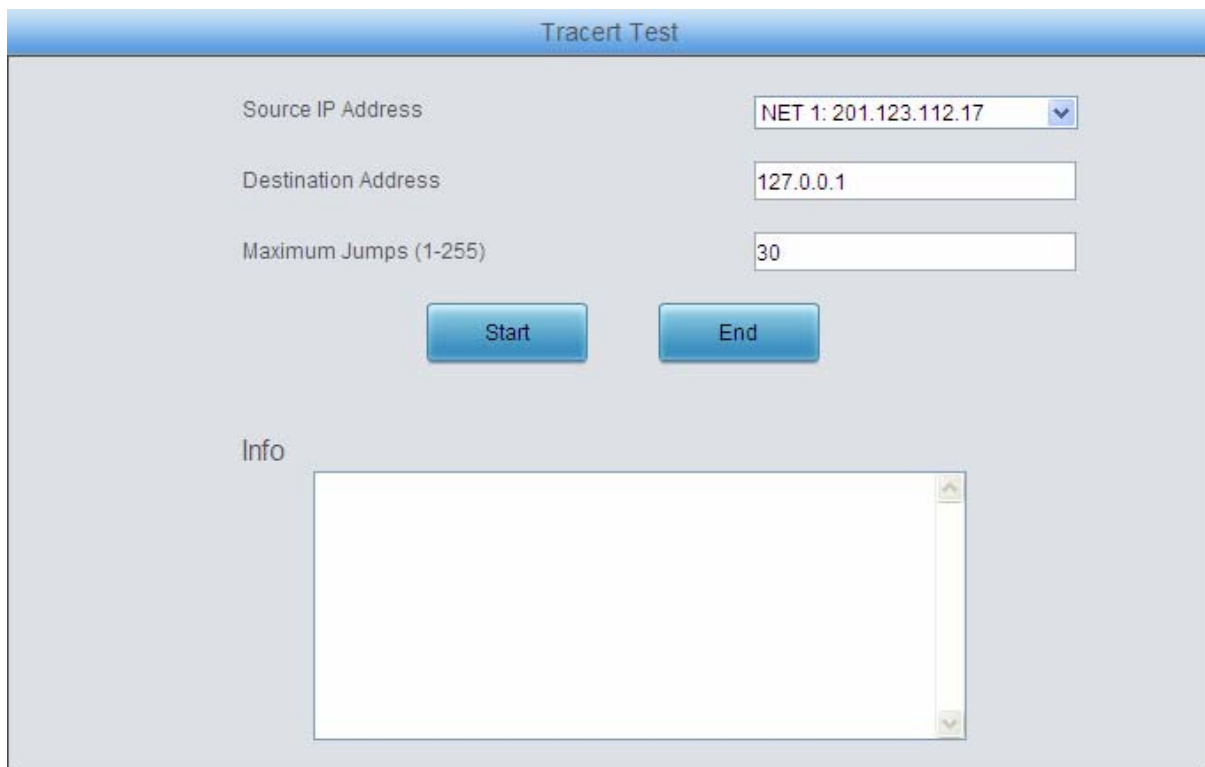
The image shows a web-based interface titled "Tracert Test". It features three input fields: "Source IP Address" with a dropdown menu showing "NET 1: 201.123.112.17", "Destination Address" with a text box containing "127.0.0.1", and "Maximum Jumps (1-255)" with a text box containing "30". Below these fields are two blue buttons labeled "Start" and "End". At the bottom, there is an "Info" label next to a large, empty rectangular box with a vertical scrollbar on the right side.

Figure 3-57 Tracert Test Interface

See Figure 3-57 for the Tracert Test interface. A Tracert test can be initiated from the gateway on a designated IP address to check the routing status between them. The table below explains the configuration items shown in the above figure.

Item	Description
Source IP Address	Source IP address where the Tracert test is initiated.
Destination Address	Destination IP address on which the Tracert test is executed.
Maximum Jumps	Maximum number of jumps between the gateway and the destination address, which can be returned in the Tracert test. Range of value: 1~255.
Info	The information returned during the Tracert test, helping you to learn the detailed information about the jumps between the gateway and the destination address.

After configuration, click **Start** to execute the Tracert test; click **End** to terminate it immediately.

3.7.10 Modification Record



Figure 3-58 Modification Interface

The Modification Record interface is used to check the modification record on the web configuration. Click **Check** and the modification record will be shown on the dialog box. See Figure 3-58. Click **Download** to download the record file.

3.7.11 Backup & Upload

The screenshot shows two sections: 'Data Backup' and 'Data Upload'. The 'Data Backup' section has a dropdown menu set to 'Configuration file', a text instruction 'Click the 'Backup' button on the right to backup the file.', and a 'Backup' button. The 'Data Upload' section has a dropdown menu set to 'Configuration file', a text input field, a 'Browse...' button, and an 'Upload' button. A red note at the bottom states: 'Note: Please edit the CSV file with Notepad.'

Figure 3-59 Backup & Upload Interface

See Figure 3-59 for the Backup and Upload interface. To back up data to your PC, you shall first choose the file in the pull-down list and then click **Backup** to start. To upload a file to the gateway, you shall first choose the file type in the pull-down list, then select it via **Browse...**, and at last click **Upload**. The gateway will automatically apply the uploaded data to overwrite the current configurations.

3.7.12 Factory Reset

The screenshot shows the 'Factory Reset' section with a text instruction 'Click the button 'Reset' below to restore to factory settings.' and a 'Reset' button.

Figure 3-60 Factory Reset Interface

See Figure 3-60 for the Factory Reset interface. Click **Reset** to restore all configurations on the gateway to factory settings.

3.7.13 Upgrade

Current Version	
Serial Number	000001861
WEB	1.6.2_2015082418
Service	1.6.2_2015082418
Uboot	2.0.9_201505
Kernel	#351 SMP Wed Aug 5 10:15:28 CST 2015
Firmware	16

Select an Update File

Figure 3-61 Upgrade Interface

See Figure 3-61 for the upgrade interface where you can upgrade the WEB, gateway service, kernel and firmware to new versions. Select the upgrade package “*.tar.gz” via **Browse...** and click **Update** (The gateway will do MD5 verification before upgrading and will not start to upgrade until it passes the verification). Wait for a while and the gateway will finish the upgrade automatically. Note that clicking **Reset** can only delete the selected update file but not cancel the operation of **Update**.

3.7.14 Change Password

Change Password	
Current Username	<input type="text" value="admin"/>
Current Password	<input type="password"/>
New Username	<input type="text"/>
New Password	<input type="password"/>
Confirm New password	<input type="password"/>

Note: The username and the password can consist only of numbers, letters or the underline.

Figure 3-62 Password Changing Interface

See Figure 3-62 for the Password Changing interface where you can change username and password of the gateway. Enter the current password, the new username and password, and then confirm the new password. After configuration, click **Save** to apply the new username and password or click **Reset** to restore the configurations. After changing the username and password, you are required to log in again.

3.7.15 Device Lock



Figure 3-63 Device Lock Configuration Interface

See Figure 3-63 for the Device Lock Configuration interface. You can select at least one item as the condition to judge whether to lock the gateway or not, that is, as long as an item in the selected list is modified, the gateway will be locked. You shall enter the password which is necessary for device unlock. After your setting, click **Lock** and the device lock interface will be locked. See Figure 3-64. To unlock the interface, enter your password and click the **Unlock** button.



Figure 3-64 Unlock Device Interface

As long as an item in the selected list in Figure 3-64 is modified, the gateway will be locked. See Figure 3-65. In such case, only five pages including *system info*, *network setting*, *change password*, *device lock* and *restart* are available. Calls will all be rejected. Enter the device unlock interface (Figure 3-64) and input your password to unlock the device.

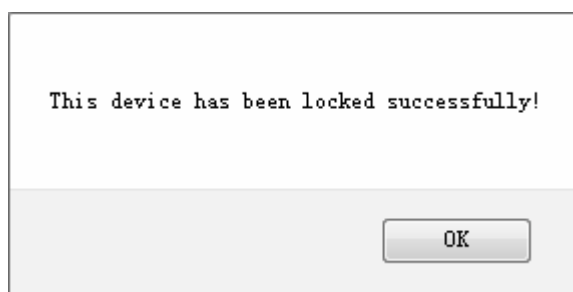


Figure 3-65 Device Lock Interface

3.7.16 Restart



Figure 3-66 Service/System Restart Interface

See Figure 3-66 for the Restart interface. Click **Restart** on the service restart interface to restart the gateway service or click **Restart** on the system restart interface to restart the whole gateway system.

Chapter 4 Typical Applications

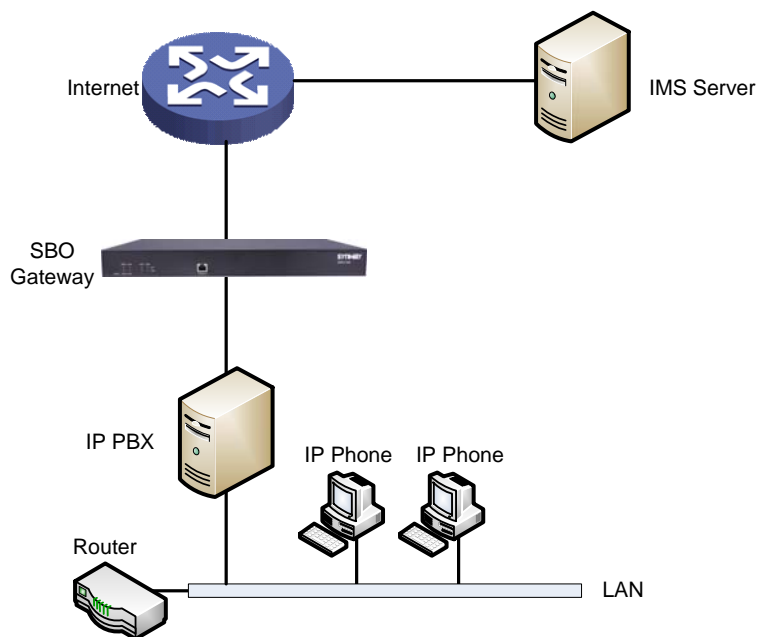


Figure 4-1 Application 1

1. Configure SIP Settings.

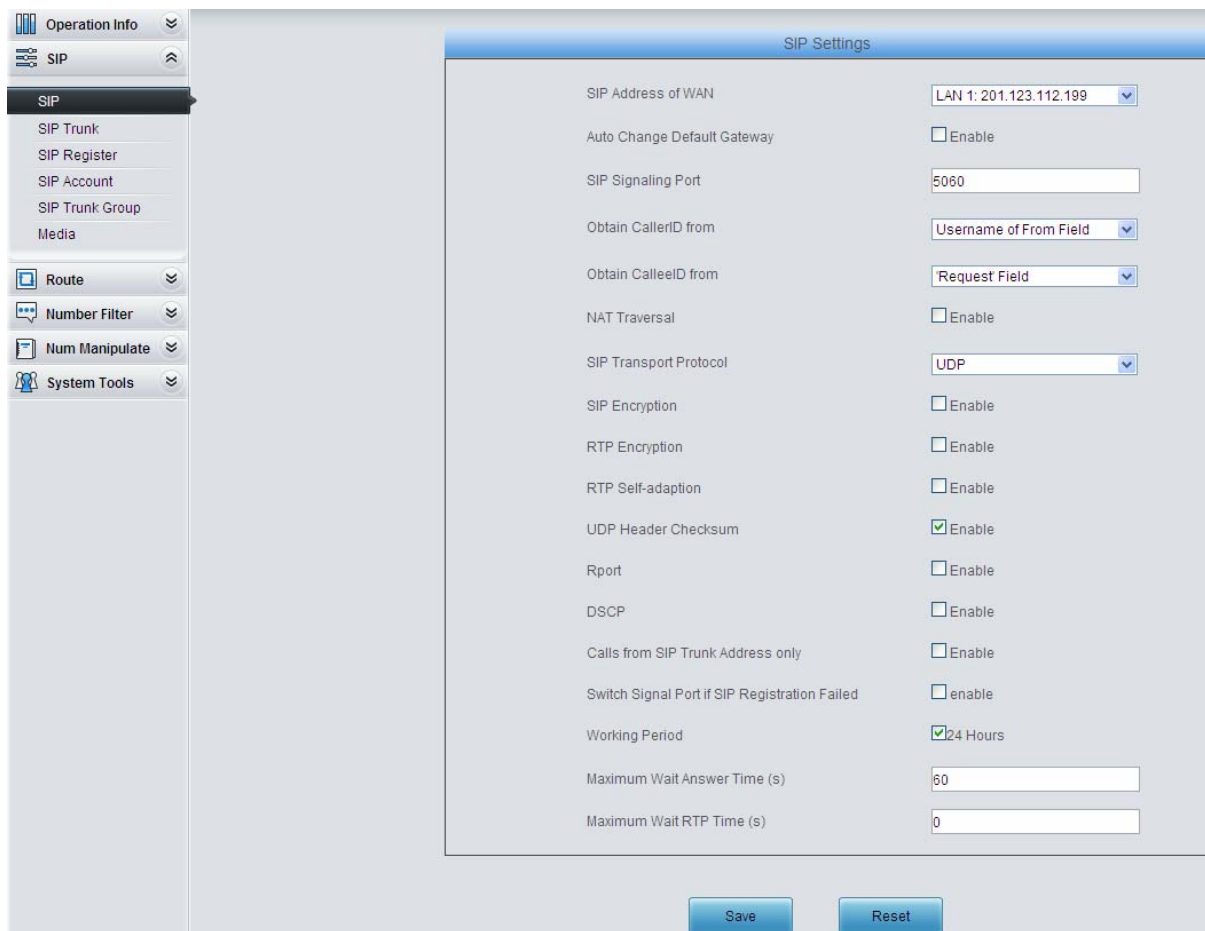


Figure 4-2

2. Add the IP addresses of the SIP terminal.

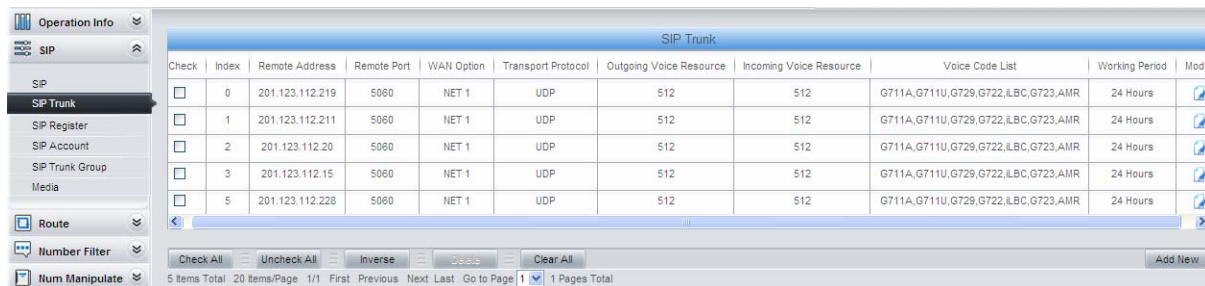


Figure 4-3

3. Add the SIP trunks into the corresponding SIP trunk groups.

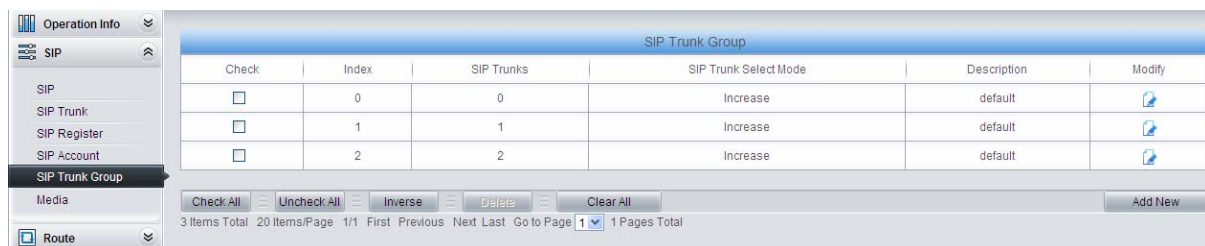


Figure 4-4

4. Set routing parameters. You may adopt the default value 'Route before Number Manipulate' for both configuration items.



Figure 4-5

5. Set IP→IP routing rules to route calls from different SIP trunk groups to the corresponding SIP trunk groups.



Figure 4-6

6. Set number manipulation rules. When the gateway receives a call from IP, it will first check the CalleeID prefix. If the CalleeID prefix is 7 or 8, the gateway will delete it before routing the call to the corresponding SIP trunk group.

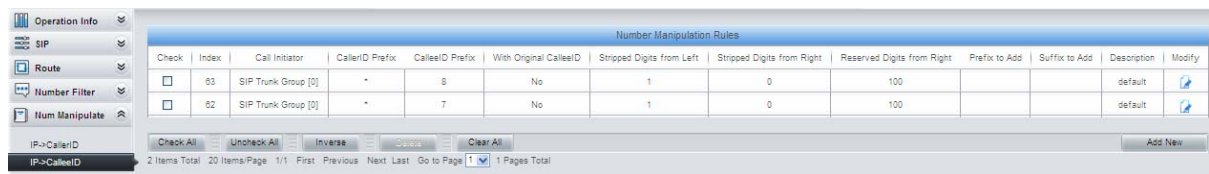


Figure 4-7

Appendix A Technical Specifications

Dimensions

440×44×267 mm³

Weight

About 3.1 kg

Environment

Operating temperature: 0℃—45℃

Storage temperature: -20℃—85℃

Humidity: 8%— 90% non-condensing

Storage humidity: 8%— 90% non-condensing

NET

Amount: 2 (10/100/1000 BASE-TX (RJ-45))

Self-adaptive bandwidth supported

Auto MDI/MDIX supported

Console Port

Amount: 1 (RS-232)

Baud rate: 115200bps

Connector: RJ45 (See [Hardware Description](#) for signal definition)

Data bits: 8 bits

Stop bit: 1 bit

Parity unsupported

Flow control unsupported

Note: Follow the above settings to configure the console port; or it may work abnormally.

Power Requirements

Input power: 100~240V AC

Maximum power consumption: ≤22W

Signaling & Protocol

SIP signaling: SIP V1.0/2.0, RFC3261

Audio Encoding & Decoding

G.711A 64 kbps

G.711U 64 kbps

G.729 8 kbps

G.723 5.3/6.3 kbps

G.722 64 kbps

AMR 4.75/5.15/5.90/6.70/7.40/7.95/10.20/12.20 kbps

iLBC 15.2 kbps

Sampling Rate

8kHz

Safety

Lightning resistance: Level 4

Appendix B Troubleshooting

1. What to do if I forget the IP address of the SBO gateway?

Long press the Reset button on the gateway to restore to factory settings. Thus the IP address will be restored to its default value:

NET1: 192.168.1.101

NET2: 192.168.0.101

2. In what cases can I conclude that the SBO gateway is abnormal and turn to Synway's technicians for help?

- a) During runtime, the run indicator does not flash or the alarm indicator lights up or flashes, and such error still exists even after you restart the device or restore it to factory settings.
- b) Voice problems occur during call conversation, such as that one party or both parties cannot hear the voice or the voice quality is unacceptable.

Other problems such as abnormal SIP trunk status, inaccessible calls, failed registrations and incorrect numbers are probably caused by configuration errors. We suggest you refer to [Chapter 3 WEB Configuration](#) for further examination. If you still cannot figure out or solve your problems, please feel free to contact our technicians.

3. What to do if I cannot enter the WEB interface of the SBO gateway after login?

This problem may happen on some browsers. To settle it, follow the instructions here to configure your browser. Enter 'Tools > Internet Options > Security Tab', and add the current IP address of the gateway into 'Trusted Sites'. If you change the IP address of the gateway, add your new IP address into the above settings.

Appendix C Technical/sales Support

Thank you for choosing Synway. Please contact us should you have any inquiry regarding our products. We shall do our best to help you.

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